AETA

AUDIO SYSTEMS ®

superior sound since 1978

SCOOPY+

Mono / Stereo portable audio codec
ISDN / POTS / MOBILE / IP

User Manual

Table of contents

1. Presentation – Getting started 1

1.1. Install and connect ScoopY+ 3

1.2. Audio settings 3

1.3. Select and set up network to be used: wired networks 3

1.3.1. Set up an ISDN link 3

1.3.2. Set up a (wired) IP link 4

1.3.3. Set up a POTS link 4

1.4. Select and set up network to be used: mobile networks 5

1.4.1. Set up a link in voice mode 5

1.4.2. Set up a (mobile) IP link 5

2. Functions 6

2.1. Transmission interfaces 7

2.1.1. Ethernet/IP interface 7

2.1.2. ISDN interfaces 9

2.1.3. POTS interface 10

2.1.4. Mobile network access 11

2.1.5. Managing calls 12

2.2. Audio encoding and decoding 13

2.2.1. G711 coding 14

2.2.2. Mobile telephony coding: GSM , AMR 14

2.2.3. Mobile HD Voice coding: AMR-WB 14

2.2.4. CELP coding 14

2.2.5. G722 coding 15

2.2.6. TDAC coding 15

2.2.7. 4SB ADPCM coding 15

2.2.8. MPEG Audio Layer 2 coding 16

2.2.9. MPEG AAC algorithms 16

2.3. Audio interfaces 17

2.3.1. Analogue inputs 17

2.3.2. Mixed Inputs 18

2.3.3. Analogue Outputs 18

2.3.1. Audio monitoring 18

2.3.2. International sound 18

2.3.1. Coordination channel 19

2.3.1. USB interface 19

2.3.1. Extension interface 19

2.3.1. Recorder 19

2.3.2. Audio routing 20

2.4. Auxiliary functions 24

2.4.1. Transmission of relays 25

2.4.2. Use isolated relays 25

2.4.3. Data channel 25

2.4.4. Coordination channel 26

2.5. Supervision and control interface 27

2.5.1. “Local” control 27

2.5.2. Embedded html server: “web pages” 27
2.5.3. Additional Ethernet/IP interface 27
2.5.4. Configuration and dialling memories 27
2.5.5. Test functions 28

3. Operation 29

3.1. General principles – Control means 29
3.2. Physical description of the equipment 30
3.2.1. Front panel 30
3.2.2. Rear panel 33
3.3. Installation and set up 35
3.3.1. Mounting and connections 35
3.3.2. Initial set up 35
3.4. Initial setup of the Ethernet interface 36
3.4.1. DHCP server available 36
3.4.2. “Static” IP configuration 36
3.4.3. Checking the IP configuration 36
3.4.4. Optional configuration of the “link” Ethernet layer 37
3.4.5. Configuration of the secondary Ethernet interface 37
3.5. Managing links 38
3.5.1. Setting up and releasing links 38
3.5.2. Auto-redial feature 38
3.6. First level maintenance 39
3.6.1. Using the test loops 39

4. Detailed operating mode – User interface 40

4.1. Equipment start-up 40
4.2. Principles for the navigation 41
4.3. Dialing and text input keypad 42
4.4. Description of the menus 43
4.4.1. Network sub-menu 44
4.4.2. Algorithm sub-menu 53
4.4.3. Audio sub-menu 54
4.4.4. Profiles menu 56
4.4.5. Tools / About sub-menu 57
4.4.6. Misc sub-menu 58
4.4.7. Maintenance sub-menu 60
4.5. Setting up a link 61
4.5.1. Setting up a link in ISDN mode 61
4.5.2. Setting up links in double ISDN codec mode 63
4.5.3. Setting up a link in IP mode via Ethernet 65
4.5.4. Setting up a POTS link 68
4.5.5. Setting up a mobile telephone link 69
4.5.6. Setting up a link in IP mode via a mobile network 71
4.6. Management of the configuration profiles 74
4.7. Restricted operation mode 75
4.7.1. Principles 75
4.7.2. Locking the front panel 75
4.7.3. Unlocking the front panel 75

4.7.4. Lost the password? 76
4.8. Clearing all settings 77
4.9. Backing up and restoring the configuration 77
4.10. Displaying received SMS 77
4.11. Audio File (record) 78
4.11.1. Settings 78
4.11.2. Record 78
4.11.3. Play 78
4.11.4. Edit 79
5. Operating mode – Embedded HTML pages 80
5.1. Accessing the SCOOPY+ html pages 80
5.2. Principles of operation with html pages 81
5.3. "Status" tab 82
5.4. "Connections" tab 83
5.5. "Profiles" tab 84
5.5.1. "Remote profiles" page 84
5.5.2. "Local profiles" page 84
5.5.3. Snapshots 84
5.6. "Network" tab 85
5.6.1. "Change Network" page: default interface selection 85
5.6.2. "Ethernet Parameter" page 85
5.6.3. "AoIP Parameter" page 86
5.6.4. ISDN Parameter page 87
5.6.5. "POTS Parameters" page 87
5.6.6. "Mobile Parameter" page 88
5.7. "Audio" tab 89
5.8. "Coding" tab 91
5.9. "Misc" tab 92
5.10. "Maintenance" tab 93
5.10.1. "Login data" page 93
5.10.2. "Tests" page 93
5.10.3. "System update" page 93
5.10.4. "Reset" page 94
5.10.5. "Event log" page 95
5.10.6. "Setup Transfer" page 96
5.11. "Alarm" tab 97
6. Technical characteristics 98
6.1. Characteristics of interfaces 98
6.1.1. Analogue audio inputs 98
6.1.2. Analogue audio outputs 98
6.1.3. Headphone outputs 98
6.1.4. Ethernet Interface 98
6.1.5. ISDN interfaces 99
6.1.6. POTS/PSTN interface 99
6.1.7. Antenna socket (mobile networks) 99
6.1.8. Batteries supply 99
6.1.9. DC power supply 100
6.2. Audio performance 101

6.2.1. Transmission gain 101
6.2.2. Amplitude-frequency response 101
6.2.3. Group delay distortion 101
6.2.4. Idle channel noise 101
6.2.5. Total distortion vs frequency and level 102
6.2.6. Crosstalk 102
6.2.7. Gain and phase difference between channels 102
6.3. Network protocols and ports 103
6.4. Dimensions and weight 103
6.5. Environmental characteristics 103
6.6. Options 104
6.6.1. Network options 104
6.6.2. Other options 104
6.7. Accessories and related products 105
7. Annexes 107
7.1. Additional information on the algorithms and protocols used 107
7.1.1. Auxiliary data in the MPEG frames 107
7.1.2. Reed-Solomon encoding 107
7.1.3. H221 framing 107
7.2. Overview of the SIP protocol 108
7.2.1. What is SIP? 108
7.2.2. Setting a link with SIP 108
7.2.3. Setting a link without a SIP server 109
7.3. Some methods to deal with NAT routers and firewalls 110
7.3.1. Links via a private network 110
7.3.2. Links through a public network (Internet) 110
7.3.3. Summary and reminder of essential rules 114
7.4. Notice regarding open source code 115
8. Index 116

Base on Scoopy+ Version 3.0

1. Presentation – Getting started

The SCOOPY+ codec allows the bi-directional transmission of one or two audio signals with bit rate reduction, over various transmission media: ISDN lines, PSTN telephone lines, IP protocol networks…

The standard version of the codec includes an Ethernet interface for IP transmission. The unit can be complemented with many options providing additional network interfaces, coding algorithms, etc.

One outstanding feature of AETA codecs in ISDN mode is the 5A System®: on receiving an incoming ISDN call, the unit can automatically detect the coding algorithm and parameters of the calling codec, and then adjust itself in a compatible configuration so that the connection succeeds regardless of the initial configuration and that of the remote unit.

In IP mode, the codec features the same ease of operation thanks to the use of the SIP and SDP protocols.

This chapter gives **basic instructions** for a quick start. It obviously does not provide all the information for full control. For comprehensive information one can refer to the rest of this manual:

- Chapter 2 describes all the **functions** and features of the SCOOPY+ (but not necessarily with all the operating modes)
- Chapter 3 gives a **physical description** of the unit, shows its **setting up** and **operation principles**.
- Chapter 4 details **menus** and **operating modes**.
- Chapter 4.11 deals with using the **html server** embedded in SCOOPY+
- Chapter 6 provides all the **technical characteristics** of the SCOOPY+
- The annexes bring miscellaneous additional information, including an **index** you can use to look for a given information topic.

The following table shows the main features of the product. Functions marked with ● in this table are available as options.

® 5AS = Aeta Audio Advanced Automatic Adjustment System

Characteristics		Optional
Operation modes		
Single wide band codec		
Double 7 kHz codec (ISDN mode)		●
IP transmission interface		
Ethernet Interface, 10BaseT / 100BaseT; TCP/IP, UDP/IP, RTP protocols		
Audio transmission in unicast mode: SIP signalling protocol, SDP, RTP streaming		
Audio transmission in multicast mode: RTP streaming		
Net bit rate 16 to 256 kbit/s (depending on coding algorithm)		
ISDN transmission interface		●
One S0 interface		
Single codec 64 or 128kbit/s , or double codec 64 kbit/s		
5AS auto configuration on incoming calls		
Transmission interface on PSTN telephone line		●
One "2 wire" telephone interface		
« POTS codec » mode with integrated V34 modem and CELP coding, 12 to 24 kbit/s		
Mobile network access		●
Integrated 2G/3G/3G+/LTE (depending on version) network access module with one internal antenna (1 sockets for get out the antenna)		
Voice mode: standard telephone or "HD Voice" (7 kHz with AMR-WB)		
Packet data mode: IP protocol, SIP signalling, SDP, RTP streaming, net bit rate 16 to 256 kbit/s (depending on coding algorithm)		
External 3G/LTE module connection via USB socket (data mode only)		
SMS reception		
Audio coding algorithms	(audio modes)	
G711 (standard telephone)	Mono	
GSM, AMR (mobile telephone)	Mono	
AMR-WB / G722.2 (mobile "HD Voice"	Mono	
G722 SRT, H221, H242	Mono	
CELP 7 kHz	Mono	
MPEG Audio Layer II	Mono, Stereo, Double mono, Joint stereo	
MPEG AAC-LC , HE-AAC, HE-AAC v2	Mono, Stereo	●
4 sub-band ADPCM (low latency)	Mono	
TDAC (ISDN mode only)	Mono	●
Audio interfaces		
3 analog MIC/line inputs and two analog outputs with adjustable gain		
Level display for encoder inputs and decoder outputs		
2 stereo headphone sockets for monitoring, balanced send/receive		
Auxiliary functions (available depending on transmission interface)		
Relay management : 2 keys and 2 LEDs		
Data channel 300 to 9600 bauds		
Audio coordination channel with Mobile option (up to 7kHz)		
Control and supervision		
Keyboard and LCD display on front panel		
Programmable set-up/dial memories		
Ethernet/IP remote control		
Embedded html server		
Recording		●

Table 1 – Main features of the SCOOPY+

1.1. Install and connect ScoopY+

- Plug on a power source or put 6 C batteries.
- Turn SCOOPY + on by holding Esc for >3 seconds. (Hold down Esc for > 3 seconds to turn it off!)
- Connect the necessary audio interfaces *(details: page 17)*
- Connect ScoopY+ on the transmission network *(details: page 33 and following)*
- Using the ScoopY+ menus: Esc key or OK key to activate the main menu, use the joystick to select a sub-menu or a parameter, enter or validate with the OK key. Esc jumps back up to the previous menu level, until getting back to the base screen (with level display). *(details: page 41)*

1.2. Audio settings

- With factory settings, all inputs are routed to the main program, and clipping level is set at +12 dBu for both inputs and outputs.
- Enter the menus using OK, select **Setup** then **Audio**. Perform necessary settings, *see details further on page 54*.

1.3. Select and set up network to be used: wired networks

- Select network: **Setup** menu then **Network**, then select **Change Network**, OK. In the proposed choice, select the desired network (**ISDN**, **MOBILE**, **Ethernet**, **POTS**). Validate with OK.
- Select the audio coding: menu **Setup** / **Algorithm**, then **Other**, then OK. Browse the available choice with the arrows, and make a selection with OK. Restart the same procedure to change for another coding setup.
  ⓘ *The available choice depends on the transmission network! For more details on coding, see page 13.*

1.3.1. Set up an ISDN link

- If needed, select the protocol with **Setup** / **Network** / **ISDN Parameter** / **Protocol**. *More details: see page 48.*
- Enter the remote number to dial, using the keypad, and press the key.
- If more than one B channel is involved due to the coding algorithm used, you must enter a second number, then , and so on. If the last dialled number is adequate, just confirm by pressing without typing a number again.
- Hang up with the key to release the connection (you must confirm by pressing again).

1.3.2. Set up a (wired) IP link

Over a public IP network, and especially when no SIP server is used, it is highly recommended to use a STUN server.
The address of a STUN server can be set in the SCOOPY+ html pages (*see page 86*) or via the menu: **Setup** / **Network** / **AoIP Parameter** / **STUN Server**, enter the address of a STUN server (*we propose our server* ***stun.aeta-audio.com****, look also the support pages on our web site* www.aeta-audio.com). Enable or disable STUN with **Setup** / **Network** / **AoIP Parameter** / **STUN Mode** (**On** or **Off**).
More details: see page 111.

- Check the Ethernet interface is active thanks to the LED on the Ethernet socket on the back, and check an IP address is allocated: menu **Tools** / **About** / **Local IP**.
- The default setting uses a DHCP server to get an IP address, which is suitable for most occasions. *In other situations, look for more details on page 36.*
- Using the keypad, enter the remote number to call (numeric IP address, or SIP URI if a SIP server is used), then press the key.
- Hang up with the key (you must confirm by pressing again).
- *When using a SIP server, some data must be entered beforehand using the* ***AoIP Parameter*** *menu; for more details, refer to page 46.*

1.3.3. Set up a POTS link

- For connecting the line, you must plug the POTS line in ANALOG socket.
- If needed, adjust the POTS line parameters using **Setup** / **Network** / **POTS Parameter**. *Details on these settings: see page 49.*
- Enter the remote number to dial, using the keypad, and press the key.
- Hang up with the key to release the connection (you must confirm by pressing again).

1.4. Select and set up network to be used: mobile networks

To set links over a mobile network, you must have a SIM card with a subscription suitable for the use. Specifically, for an IP mode transmission the subscription must include access to data transmission, and RTP audio streams must be allowed.

- While the unit is switched off, insert the SIM card into the drawer on the rear side of the SCOOPY+.
- Switch on the SCOOPY+ (depress ⏻ a few seconds).
- Enter the menu **Setup** / **Network** / **Other (Network)** / **Mobile (Int.)** / **Mobile Parameter** / **PIN**. Enter the PIN code for the SIM card using the keypad then OK.

1.4.1. Set up a link in voice mode

This mode allows communicating with any telephone terminal through the regular telephone service. It also allows to benefit from the 7 kHz wide band service known as "HD Voice" whenever the remote terminal is compatible and the network supports the service.

- In the menu **Setup** / **Network** / **Mobile Parameter**, select **Mode** / **Cellphone**
- Afterwards, go to the menu **Setup** / **Network** / **Mobile Parameter** / **Network Settings** / **Preferred Technology** / **Auto**
- The **Network Select** menu enables you to select among the available operators, if your mobile subscription entiltes you to do so.
- Enter the remote number to dial, using the keypad, and press the ✆ key.
- Hang up with the ✆ key to release the connection (you must confirm by pressing ✆ again).

1.4.2. Set up a (mobile) IP link

⚠ Over a public IP network, and especially when no SIP server is used, it is highly recommended to use a STUN server.
The address of a STUN server can be set in the SCOOPY+ html pages (*see page 86*) or via the menu: **Setup** / **Network** / **AoIP Parameter** / **STUN Server**, enter the address of a STUN server (*we propose our server* ***stun.aeta-audio.com****, look also the support pages on our web site* www.aeta-audio.com). Enable or disable STUN with **Setup** / **Network** / **AoIP Parameter** / **STUN Mode** (**On** or **Off**).
More details: see page 111.

- In the menu **Setup** / **Network** / **Mobile Parameter**, select **Mode** / **IP Mode**
- Come back to the menu **Setup** / **Network** / **Mobile Parameter** / **Network Settings** / **Preferred Technology** / **Auto**
- Go to the menu **PS Settings** / **APN**: enter the operator's APN code, using the keypad, then press OK.
- Come back to the base screen using the Esc key.
- Using the keypad, enter the remote number to call (numeric IP address, or SIP URI if a SIP server is used), then press the ✆ key.
- Hang up with the ✆ key (you must confirm by pressing ✆ again).
- *When using a SIP server, some data must be entered beforehand using the* ***AoIP Parameter*** *menu; for more details, refer to page 46.*

2. Functions

The following synoptic diagram shows the basic functions of the equipment.

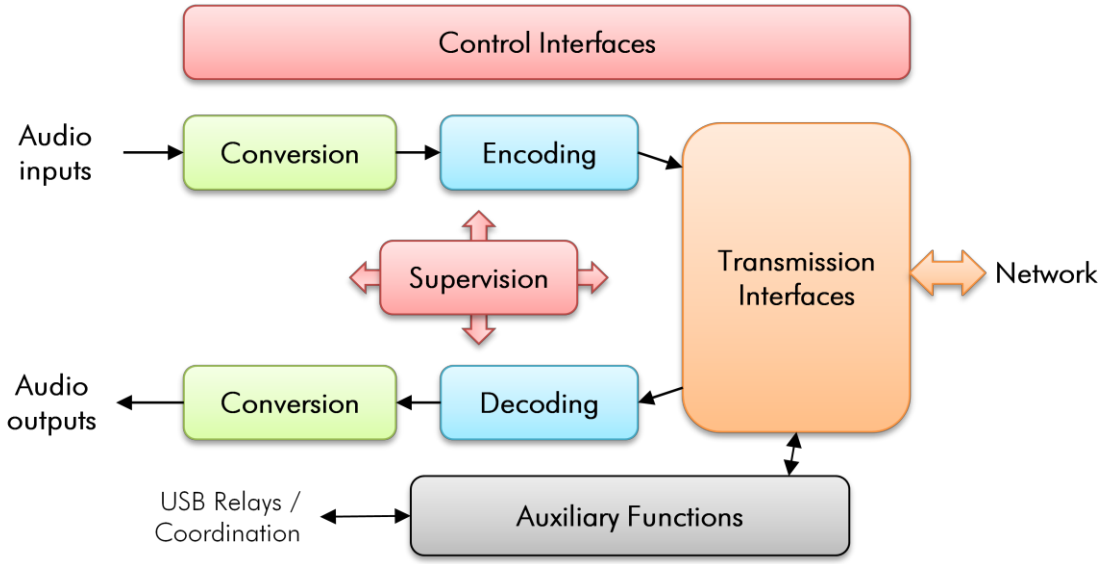


Figure 1 – Functional diagram of the equipment

The audio signals to be transmitted are converted (when needed) to digital format, then the encoding function reduces the bit rate, using a selectable algorithm; the resulting bit flow is sent to one of the available transmission interfaces: Ethernet interface, ISDN interface, PSTN interface, mobile network...

The transmission interface functional block also extracts compressed data coming from the network and sends them to a decoding block that reproduces uncompressed audio data. Last, the audio signals are output to analogue outputs.

Monitoring the audio interfaces is possible thanks to headphones and level meters for the inputs/outputs.

In addition to the main task of transmitting an audio programme, the SCOOPY+ can also transmit auxiliary information, usually by embedding them inside the transmitted audio streams.

Supervision and controlling the unit is performed using various remote control interfaces, and of course by means of the displays and controls on the front panel.

2.1. Transmission interfaces

The SCOOPY+ features in all versions an Ethernet interface for IP protocol networks.

One ISDN, PSTN/POTS and mobile network access is optionally available.

2.1.1. Ethernet/IP interface

The IP interface is a 10BaseT/100BaseT Ethernet interface allowing transmission of the audio programmes in a wide range of possible bit rates. The audio stream is always transported under the RTP/UDP protocol.

IP unicast mode

The most classical transmission mode is unicast: audio connection with one distant device, generally bidirectional. This mode can be used on all types of networks links, LAN or WAN, including links via Internet. The SCOOPY+ implements the SIP protocol, which allows it to interoperate with IP phones and other SIP compatible audio codecs, in a way similar to ISDN or POTS connections. Links can be set up in two ways:

- "Peer to peer" connection between two compatible units
- Use of a SIP proxy server to set up the link, or a SIP PBX

Details about the SIP protocols can be found in the annex (see 7.2, Overview of the SIP protocol).

The audio coding algorithm can be selected depending on the required quality and the available network bandwidth. The following algorithms are currently available:

Codec	Bit rate (coding)	Bit rate (total)[1]	Audio bandwidth	Typical use, main features
G711	64 kbit/s	86 kbit/s	3 kHz	Voice, telephony Compatible with IP phones
CELP	24 kbit/s	28,5 kbit/s	7 kHz	Suitable for high quality speech; Low network bandwidth consumption
G722	64 kbit/s	86 kbit/s	7 kHz	High quality speech. Compatible with some IP phones.
MPEG Layer II	64 to 256 kbit/s	73 to 275 kbit/s	Up to 20 kHz	Highest quality, suitable for speech and music
MPEG AAC-LC	16 to 256 kbit/s	30 to 277 kbit/s	Up to 20 kHz	Low bit rate, suitable for speech and music
MPEG HE-AAC and HE-AAC v2	16 to 128 kbit/s	23 to 139 kbit/s	Up to 20 kHz	Very low bit rate, suitable for speech and music
4SB ADPCM	128 or 256 kbit/s	173 or 301 kbit/s	15 kHz	Low latency, suitable for speech and music

Table 2 – Overview of algorithms available in IP mode

[1] Informative value; higher than the "net" encoded audio bit rate because of the protocol overhead

SIP and SDP protocols

The SIP protocol is a signalling protocol, used for IP connections, which allows the SCOOPY+ to interoperate with IP phones and other SIP compatible audio codecs, in a way similar to ISDN or POTS connections. Details about the SIP protocols can be found in the annex (refer to 7.2, Overview of the SIP protocol).

One significant advantage is the inclusion of SDP, a protocol which allows the connecting devices to automatically negotiate and agree on the coding profile to use. Thanks to this system, it is not necessary to set the units in the same way before setting up a connection. Moreover, the calling party needs not know how the remote unit is configured before initiating a link.

Note: the SIP protocol does not mandatorily imply the use of a server. Codecs can set up point-to-point links using this protocol, and benefit from some its advantages.

Packet replication

SCOOPY+ also proposes an RTP transmission mode with enhanced reliability, using packet replication. When enabling this mode, every packet is transmitted twice; with such system a lost packet has no effect since the receiver still gets the other copy of the packet. In this way, stable links can be obtained even with a high packet loss rate. Of course, as a disadvantage the bit rate is double; you must make sure this stays compatible with the transmission medium.

Remote control via IP

In addition, the Ethernet interface can be used for configuring or remote controlling the unit, with two control methods:

- SCOOPY+ provides html pages which allow to get complete control over the unit using a web browser, via port 80 (default port for http protocol). See in chapter 4.11 the detailed operating mode.
- TCP port 6000 can be used for “command line” control, suitable for codec supervision software such as Codec Live, MDC.net, etc.

2.1.2. ISDN interfaces

For access to the ISDN, the transmission interface is one S0 BRI (Basic Rate Interface), for transmission over one to two 64 kbit/s B channels.

The codec synchronises itself onto the ISDN network clock when a link is active.

Network protocols

Available protocols:

- "Euro ISDN" (or ETSI), default protocol valid for a large number of countries, especially all over Europe.
- "NTT": valid for the Japanese network of NTT
- "NI-1": valid for numerous operators in North America. This choice is also suitable for the connection to network equipment with "NI-1" & "NI-2" protocol.

In North America (USA and Canada), the available interface is often a U0 interface (instead of S0). In such case an "NT1" network adapter must be inserted between the line and the SCOOPY+. Such adapter can be found on the local market.

5A System®

Setting an ISDN connection is often difficult, at least because of the numerous coding parameters to be set. Moreover, with most proprietary algorithms, it is mandatory for the two devices to have exactly the same settings, otherwise the connection will fail, and sometimes it is not easy to find out the reason.

5A stands for Aeta Audio Advanced Automatic Adjustment. This system makes it easier to set an ISDN connection, because the codec, on receiving a call, automatically adjusts itself, following the calling party algorithm and parameters.

When the 5A System is enabled on the unit and a call is received, the unit first detects the coding algorithm used by the calling codec, and also senses its parameters: audio mode (mono, stereo…), sampling rate, bit rate, inverse multiplexing protocol, etc. Then the unit can decode the compressed audio from the remote unit. In addition, the unit will use these same settings for encoding and sending audio to the remote unit, so that the remote unit can also decode the outgoing audio programme. The whole process just takes a few seconds. Of course, all compatible coding configurations can be detected automatically by the 5A System.

Note that the 5A system is only active for ISDN connections.

J52

The ITU-T J52 recommendation was defined in order to allow the interoperability of multimedia terminals over the ISDN[1], using common coding standards. It includes the following features:

- Support of ITU-T recommended coding algorithms: G711, G722, MPEG Layer II
- Framing as per ITU-T H221 recommendation, ensuring byte synchronisation and interchannel synchronisation when more than one 64 kbit/s B channel is required for the desired bit rate;
- Interoperation procedures according to ITU-T H242 recommendation;
- In the case of MPEG encoding, optional protection against transmission errors (Reed-Solomon error correction codes).

Details about MPEG and J52 can be found in the annexes (refer to 7.1, Additional information on the algorithms and protocols used).

[1] J52 is only relevant for ISDN connections

It must be noted that, thanks to the interoperation protocol, J52 codecs, when setting up a link, can negotiate automatically and agree on a configuration that is compatible with the capability of both units (regarding bit rate, channel mode, etc.). In this way, when the units differ in their capability (or make), the resulting configuration may be different from expected beforehand, but in most cases the link will work and audio will be transmitted.

As another useful consequence, this also gives users more tolerance to mistakes when configuring the units on the two sides of the transmission links, as the codecs will adapt automatically even with differences in the initial settings of the two units.

Symmetric or asymmetric coding

In most operating cases, the codec sets up **symmetrical** links, wherein the encoder and decoder use the same encoding/decoding algorithm with same settings (channel mode, etc.). In other words, the link is "full-duplex" and the same encoding/decoding type is used on both directions.

For a link over the ISDN, in fact there are cases when the link is **asymmetric**, with a different coding mode in each direction. This may happen in some cases with the J52 protocol. To give some examples, it is possible to send MPEG Layer II in one direction and receive G722 in the reverse direction, or send MPEG stereo and receive MPEG mono, etc.

2.1.3. POTS interface

The interface is a "two wire" analog telephone access, with characteristics adjustable depending on the country. Dialling normally uses DTMF, but for older switching equipment it is possible to use pulse dialling.

SCOOPY+ includes a V34 modem which transmits via this line a bidirectional audio flow, encoded at a nominal 24 kbit/s bit rate. Depending on the line quality and the quality of the link with the remote codec, this bit rate is automatically negotiated and dynamically adjusted from 12 to 24 kbit/s.

A "protected" mode can be activated, which increases the resilience to transmission errors, at the cost of a higher latency (encoder to decoder delay). *You must make sure to set this parameter the same way on both devices / both ends of the link.*

2.1.4. Mobile network access

Units equipped with the "Wireless" option include an integrated module for access to 2G/3G/3G+ mobile networks, and a holder for a SIM card.

Depending on the version the accessible networks are 2G (GSM, EDGE), 3G (UMTS), 3G+ (HSDPA, HSUPA, HPA…), and 4G/LTE.

The SCOOPY+ include one or two internal antennas (4G/LTE module only). If an external antenna is necessary, you can connect one multiband antenna (to be selected for compliance with the mobile network characteristics) on SCOOPY+.

Lastly, SCOOPY+ can display the received SMS messages.

Mobile voice mode – HD Voice

The integrated module allows using the mobile phone service, for communicating with all ISDN or PSTN telephone terminals or hybrids, or with other mobile terminals. The quality is in such case that of mobile connections, with a 300-3400 Hz bandwidth and coding such as GSM, EFR, AMR...

Now many mobile networks also propose "HD Voice", an extension of this mobile telephone service. With this new capability, compatible terminals implement the AMR-WB coding algorithm (standardised as G722.2 by the ITU-T) and provide speech transmission with a 50-7000 Hz bandwidth and a quality very similar to the well-known G722. Automatic fallback to the standard coding takes place if the network does not support the service or one of the terminals does not feature this capability.

No special subscription, other than to the regular telephone service, is needed, but for most operators only the 3G/3G+ base stations support the service.

ⓘ *This sometimes makes people believe that HD Voice is related to the mobile IP service, but this is definitely not the case.*

More and more mobile phones now support this service, especially (but not only, and not all) smartphones. All AETA codecs in "Wireless" version support HD Voice, namely:

- SCOOP 4+ in "wireless" version
- SCOOP 5+ in "wireless" version
- ScoopFone HD

Mobile IP mode

The other service available with mobile access is the data packet transmission mode, abbreviated as "PS" (for Packet Switched), with IP protocol.

This mode brings similar capabilities as a wired IP connection via the Ethernet interface, as described above in 2.1.1, with some distinctive characteristics:

- This requires a subscription including access to the data service, with conditions compliant with the application. Among other requirements, an APN (Access Point Name) must be provided that allows this type of media stream.
- The available bit rate depends on various factors; first the network technology (2G/3G/3G+…), but also the traffic level in the radio cell, the operator's network capacity, possibly the type of subscription. This may bring on restrictions for the usable compression algorithms.
- The multicast mode is not available on mobile networks
- Setting a link implies first activating the data connection, before actually initiating an audio stream transmission link

Using an external USB module

Instead of the integrated module, it is possible to plug a USB mobile module or "key", in order to access mobile IP transmission, with more or less similar conditions as described above.

ⓘ *However be aware:*

- This capability is optional
- The USB module must be from the list of devices supported by AETA. *As this list is evolving, please check our web site for up to date information.*
- The "HD Voice" mode is not available in this way
- USB devices do not feature antenna diversity

2.1.5. Managing calls

One of the transmission interfaces is selected as the *default interface* on the SCOOPY+.

A call towards a remote unit, initiated by the user of the SCOOPY+, is implicitly sent through this default interface.

On the other hand, an incoming call on any interface (regardless of the default interface) can be processed and the link established, under following conditions:

- The "called" interface must be connected and active. As an example, for mobile IP the data connection must be active.
- The codec must not be already busy with another connection.

If the call comes on an interface other than the default interface, the codec first switches to the suitable interface, and then processes the incoming call. When the link is released, it will come back to its previous state (and default interface).

2.2. Audio encoding and decoding

SCOOPY+ features a wide range of coding algorithms. Their availability depends on the transmission network used. Besides, the MPEG family algorithms feature a large configuration flexibility.

The table below synthetically describes the capabilities with the various transmission media:

Codec	Audio channels	Frequency (kHz)				Bit rate (kbit/s)	Wired			Mobile		
		16	24	32	48		PSTN	ISDN	Ethernet	3G+ LTE	UMTS	Voice
G711	Mono					64						
GSM, AMR	Mono					-						
AMR-WB	Mono					-						
CELP	Mono					12 → 22.8						
CELP	Mono					24						
G722	Mono					64						
G722-H221	Mono					64						
G722-H242	Mono					64						
TDAC	Mono					64						
4SB ADPCM	Mono					128						
4SB ADPCM	Stéréo					256						
MPEG L2	M / S					64						
MPEG L2	M / S					128						
MPEG L2	M / S					192						
MPEG L2	Stereo					256						
MPEG L2	Stereo					384						
AAC-LC	M / S					16 → 56						
AAC-LC	M / S					64						
AAC-LC	M / S					96						
AAC-LC	M / S					128						
AAC-LC	M / S					192						
AAC-LC	Stereo					256						
HE-AAC	M / S					16 → 56						
HE-AAC	M / S					64						
HE-AAC	Stereo					96						
HE-AAC	Stereo					128						
HE-AAC v2	Stereo					16 → 56						
HE-AAC v2	Stereo					64						

IP networks

Possible audio bandwidth:

3 kHz 7 kHz

15 kHz 20 kHz

Table 3 – Available coding depending on network

This table applies to the normal mode, single codec.

In double codec ISDN mode, only G711 and G722 algorithms are available for each codec.

The following chapters bring some precisions about the important features of the various algorithms and protocols available.

2.2.1. G711 coding

Application: telephony, coordination. Low latency.

G711 is the standard coding used for voice transmission on public telephone networks, and features 300 to 3400 Hz audio bandwidth. This algorithm is typically used for links over IP networks with IP telephones or VoIP gateways. Via ISDN, G711 is used for links with telephones or hybrid devices.

G711 is available only for IP or ISDN transmission

2.2.2. Mobile telephony coding: GSM , AMR

Application: telephony, mobile coordination. Moderate latency.

These algorithms are exclusively used for speech transmission over mobile telephone networks, with a 300 to 3400 Hz audio bandwidth. Gateways perform, whenever needed, transcoding in order to interface with fixed PSTN, ISDN and IP networks.

2.2.3. Mobile HD Voice coding: AMR-WB

Application: commentaries, mobile coordination. Moderate latency.

The AMR-WB coding (standardised as G722.2 by the ITU-T) is used between compatible mobile terminals, when the mobile network supports the so called "HD Voice" service, and provides speech transmission with a 50-7000 Hz bandwidth.

SCOOPY+ automatically implements this algorithm in mobile voice mode every time it is possible, and automatically falls back to standard voice coding if not (when network does not support, or the remote terminal is not compatible).

ⓘ *Unfortunately it is not possible to see directly whether AMR-WB is active or not at a given moment. You have to rely on your listening skills! However, it has to be active if the conditions are met: a) support from the network on both sides of the link, b) both terminals compatible, c) service continuity from end to end*[1]

2.2.4. CELP coding

Application: commentaries, coordination. Low capacity transmission channels

This algorithm operates in mono for a net nominal bit rate of 24 kbit/s, and provides 7 kHz bandwidth and a quality close to G722 for a much lower bit rate.

CELP is available for IP or PSTN networks. For PSTN (POTS), the bit rate may be reduced to adapt for the line quality, among the following values: 12 kbit/s, 14.4 kbit/s, 16.8 kbit/s, 19.2 kbit/s, 21.6 kbit/s, 24 kbit/s. The bandwidth is reduced accordingly.

Still for POTS lines, in addition to the normal mode two "protected modes" are available, bringing an increased resistance to transmission problems, however with an increased latency as well.

[1] At the time of writing, this requires both units to be on the same network: same operator, same country

2.2.5. G722 coding

Application: commentaries, coordination. Low latency.

This mono coding algorithm at a 64 kbit/s bit rate is a reference for commentaries, and features a 50-7000 Hz bandwidth.

It is available, ISDN, IP networks (wired or mobile).

For ISDN, three synchronisation modes are possible:

- "Statistical recovery" byte synchronisation method (alias SRT);
- H221 synchronisation; in this case, 1.6 kbit/s from the compressed data are used for this;
- H221 synchronisation and H242 protocol.

H221 synchronisation is highly recommended when possible, as it features higher reliability and faster recovery time, while degradation (because of the bit rate used for framing) is minimal.

H242 protocol, the most flexible mode, is recommended by the ITU-T, and is included in J52. However, the mode with H221 synchronisation but without H242 protocol can be useful for compatibility with old generation codecs which did not use this protocol.

No specific synchronisation is required for the IP mode.

2.2.6. TDAC coding

Application: commentaries, mono music via ISDN with only one B channel.

The SCOOPY+ can also include the TDAC algorithm. TDAC is for Time Domain Aliasing Cancellation; this is a transform coding based on an MDCT (*Modified Discrete Cosine Transform*), encoding a 15 kHz bandwidth mono signal at a 64 kbit/s bit rate.

TDAC is available as an option, only for the ISDN mode.

2.2.7. 4SB ADPCM coding

Application: commentaries, mono or stereo music. Low latency.

4SB ADPCM operates either in mono at a 128 kbit/s rate, or in stereo at 256 kbit/s, for a 15 kHz bandwidth. It features a very low latency which makes it very interesting for live duplex transmission. It also has the advantage to be very little sensitive to tandem coding.

4SB ADPCM is available for ISDN (Mono only); wired or mobile IP. However it is not recommended for mobile networks, because it requires a high bit rate and it is highly sensitive to packet losses, which can be frequent on such networks.

2.2.8. MPEG Audio Layer 2 coding

Application: mono or stereo music, high quality.

As shown on Table 3, this coding algorithm features a maximum flexibility, with many variations for bit rate, mono or stereo channel mode, sampling rates...

The two channel modes exist in three variations:

- Stereo: coding of each channel stays independent
- Dual mono: coding is similar to the previous case, but this choice applies to channel with no acoustic relationship, e.g. two languages for commentaries
- Joint stereo: applies to stereo programme, but here the encoder exploits the interchannel correlation for improved coding. To be used only for a stereo programme

The 16 and 24 kHz sampling rates feature a moderate bandwidth (respectively 7 kHz and 10 kHz) and are rather useful for commentaries.

ⓘ *The latency is rather high with these sampling rates*

MPEG L2 is available for LL, ISDN; wired or mobile IP networks.

ISDN mode specific aspects: J52

For ISDN transmission, MPEG L2 is proposed with two variations:

- "MPEG L2 J52" variation, using the ITU-T J52 protocol for the link negotiation and inverse multiplexing (B channel aggregation for connections requiring more than 64 kbit/s). Moreover, optional protection against transmission errors (FEC) is also available. For more details refer to 2.1.2, ISDN interfaces (page 9).
- MPEG L2" variation, without the J52 protocol, using a system for inverse multiplexing two B channels which is proprietary but compatible with several codecs on the market. The bit rate is limited to 128 kbit/s with this option.

2.2.9. MPEG AAC algorithms

Application: mono or stereo music, Low capacity transmission channels.

These (optional) algorithms feature a very high compression ratio, for a given audio quality, compared to Layer 2. They can operate at a sampling rate of 32 or 48 kHz, and several bit rates: 16, 20, 24, 32, 40, 48, 56, 64, 96, 128, 192, 256 kbit/s (192 & 256kbit/s only in IP mode). Three coding variations are available:

- MPEG AAC-LC ("Low Complexity"): lower compression than other variations, but lower latency.
- MPEG HE-AAC ("High Efficiency" AAC): higher compression, and the bit rate is limited to 128 kbit/s for this variation.
- MPEG HE-AAC v2 ("High Efficiency" AAC version 2): compared to the above, this coding further enhances the performance for a stereo program (not available for mono). The bit rate is limited to 128 kbit/s for this variation.

AAC codecs are available as an option for ISDN, and wired or mobile IP networks.

2.3. Audio interfaces

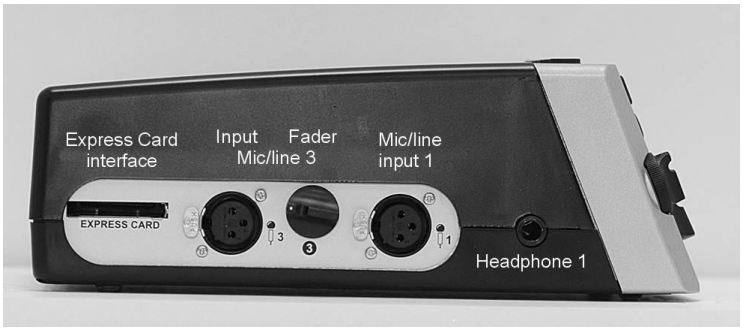


Figure 2 – Left panel

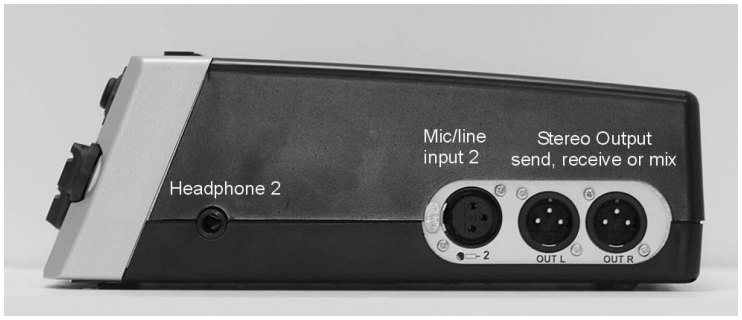


Figure 3 – Right panel

2.3.1. Analogue inputs

Scoopy+ contains an audio mixer, that enables three microphones to be mixed. The third microphone inputs accept line level via adjustable PAD, when the source is either a recorder of a mixing console. If you need more microphone inputs, you can use AETA AUDIO SYSTEMS mixer : MIXY on the extension port. Mixy will add 3 more microphones inputs plus many lines inputs. Mixy is automatically detected by the scoopy+.

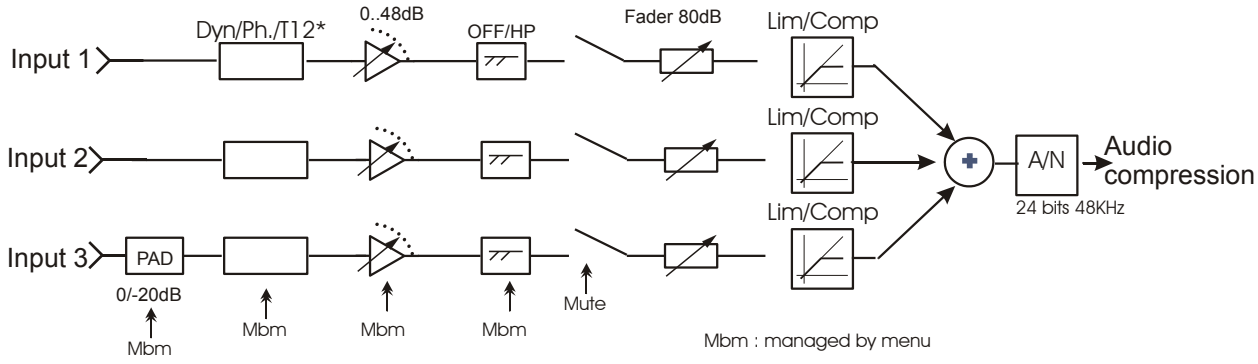


Figure 4 Mixer features

The following elements are available for each mic/line input:

- Input connector: female XLR;

- Pad switch to adjust the gain on each input
- A mute activation pad located on the front panel

By menu [1]

- Gain pad (to adjust by step of 16 dB from 0 to 48dB).
- Microphone power selection switches
  None: dynamic microphone or line live
  Phantom: phantom power supply (48V or 12V switch configuration)
  T12 : "Tonadder" (12V) for certain static microphones
- Each input has a high pass filter @ 50Hz, 18db/oct.
- Each input has a limiter preset at -8dBFs.
- Each input can be routed on main program or/and coordination line in mono or on left or right channel of stereo codec.
- Default mute configuration on power on
- PAD 20dB on the input 3 only

2.3.2. Mixed Inputs

On the Mixed signal, you have a level adjustment from -12 to +12dB by 1dB step. You can set it by menu Setup / Audio / Input / Global Gain. By default it is set to 0dB.

2.3.3. Analogue Outputs

The mixed audio signal from inputs is available on the 2 headphones and 2 output lines.
Local audio from the inputs can be mixed with the return audio signal on each headphone. The return audio signal is present in the headphone mix via the Local / Cue Mix Balance potentiometer on the front.
By default, the return program audio is assigned to the output lines. You can assign local audio program or the headphone 2 mixed audio signals to the output by menu.

2.3.1. Audio monitoring

The levels on the bar graph on the front panel of the Scoopy+ indicate the peak level of the mixed audio signal. The level displayed is registered at the analog-digital converter overloading level. The 'reference' level can be fixed by the audio menu.

2.3.2. International sound

This feature is available only with mono transmission. Each Input can be set by menu as an international sound input. Then this mode is enabled, selected inputs can be sending on the headphone right channel in place of return signal.

Warning: You can't have at the same time an international sound and coordination channel, because both use the same mixer channel.

[1] See Audio menu

2.3.1. Coordination channel

With Mobile option, you can have access to a coordination channel over Mobile network. Through menu you can assign inputs to this channel. Also, you can enable a monitoring of the coordination channel return on each headphone.

With the DYNAMIC mode, as the way of orders is not connected, the headphones still have only the program. This is when the channel order is established that the switching is done according to the choice of headphone

2.3.1. USB interface

Scoopy+ has also the capability to establish a link with a PC or a MAC. The Scoopy+ is seeing as a stereo sound card by the computer. Depending of the Scoopy+ configuration, you can record on the computer the main program (send signal on the left and return signal on the right)

On the other way, audio play on the computer is mixed with ScoopY+ inputs. See 2.3.2 for more details.

2.3.1. Extension interface

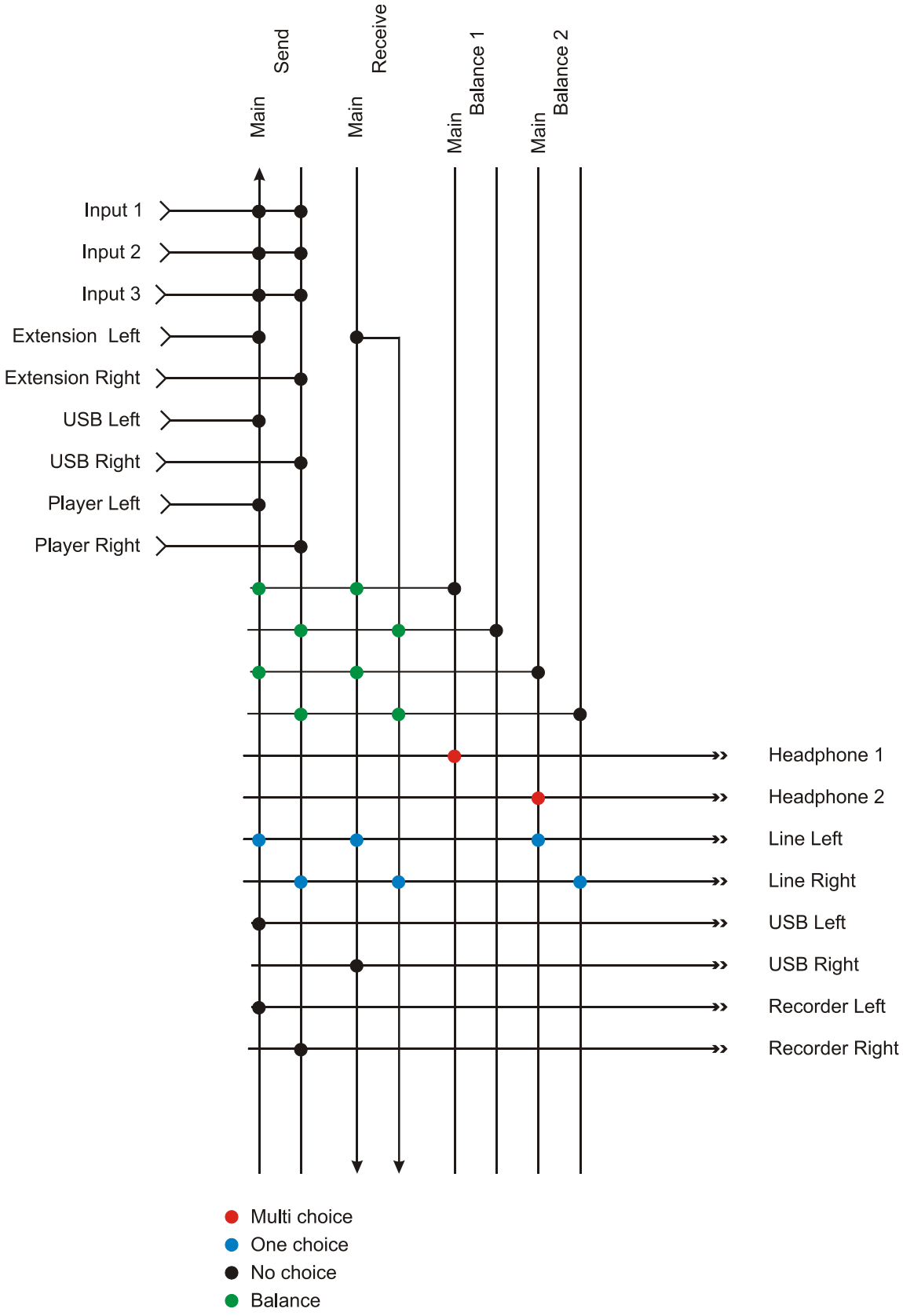
Scoopy+ has also the capability to establish a link with the small AETA 3 channels mixer (Mixy) . Scoopy+ mixes the Mixy signal with its own inputs. Like that, you can increase your configuration up to 6 microphones.

2.3.1. Recorder

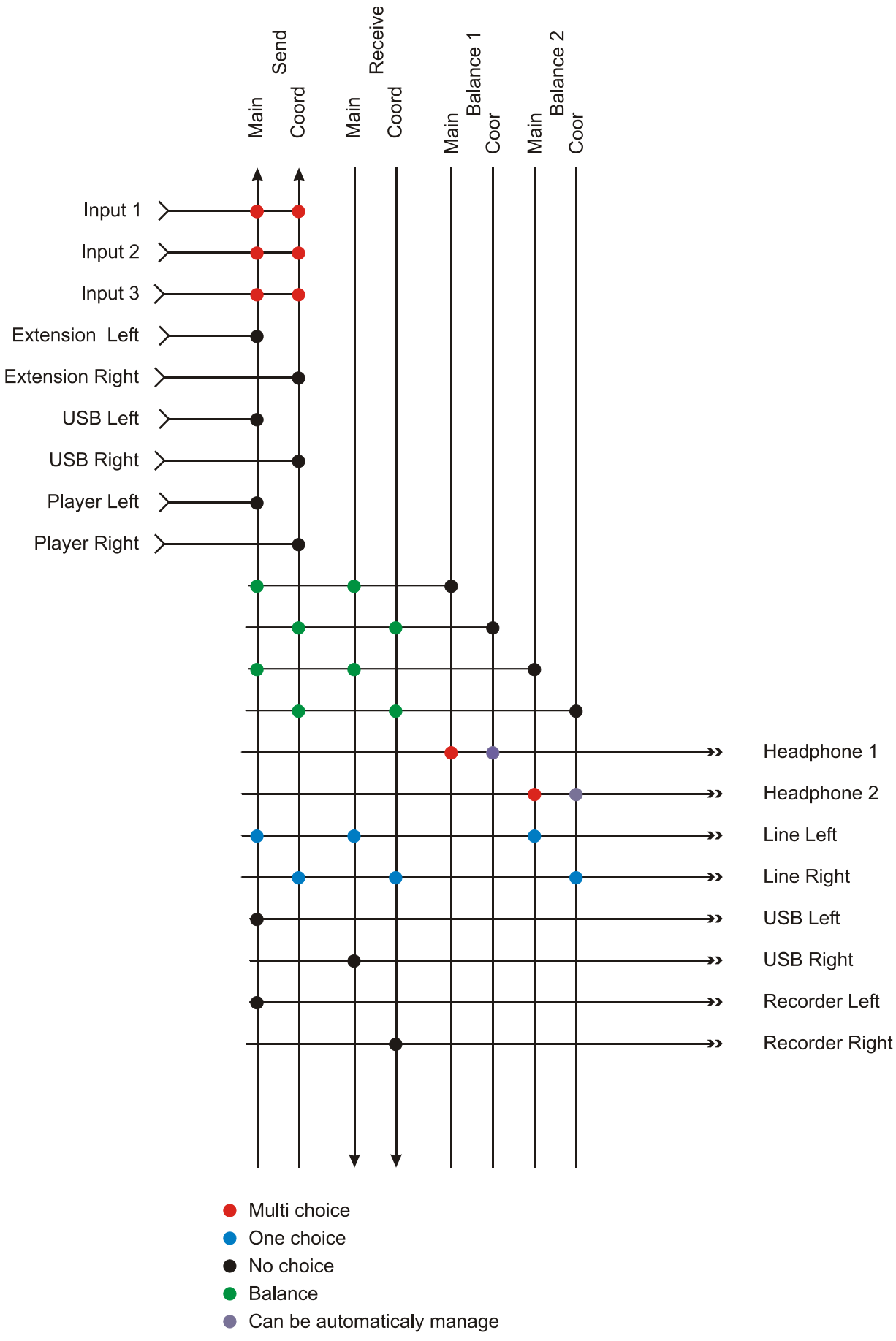
Scoopy+ can record on an SDCARD in mono or stereo the mix of inputs. See next chapter “Audio routing” and “Audio menu” for the configuration

WARNING: When you enable the coordination channel, the recorder records the main program and the coordination return channel when you select a stereo audio file. This configuration is perfect to record a remote interview.

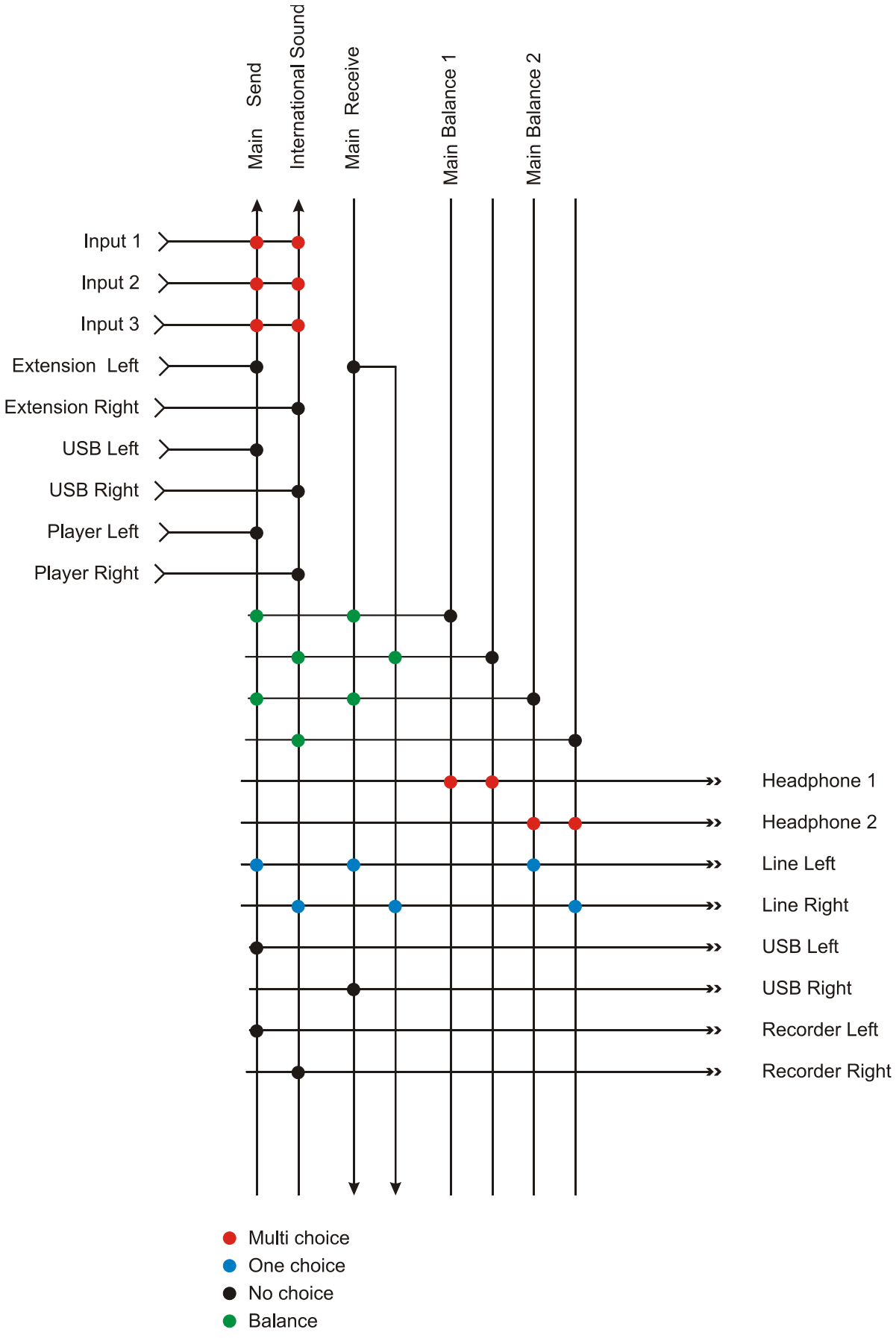
2.3.2. Audio routing



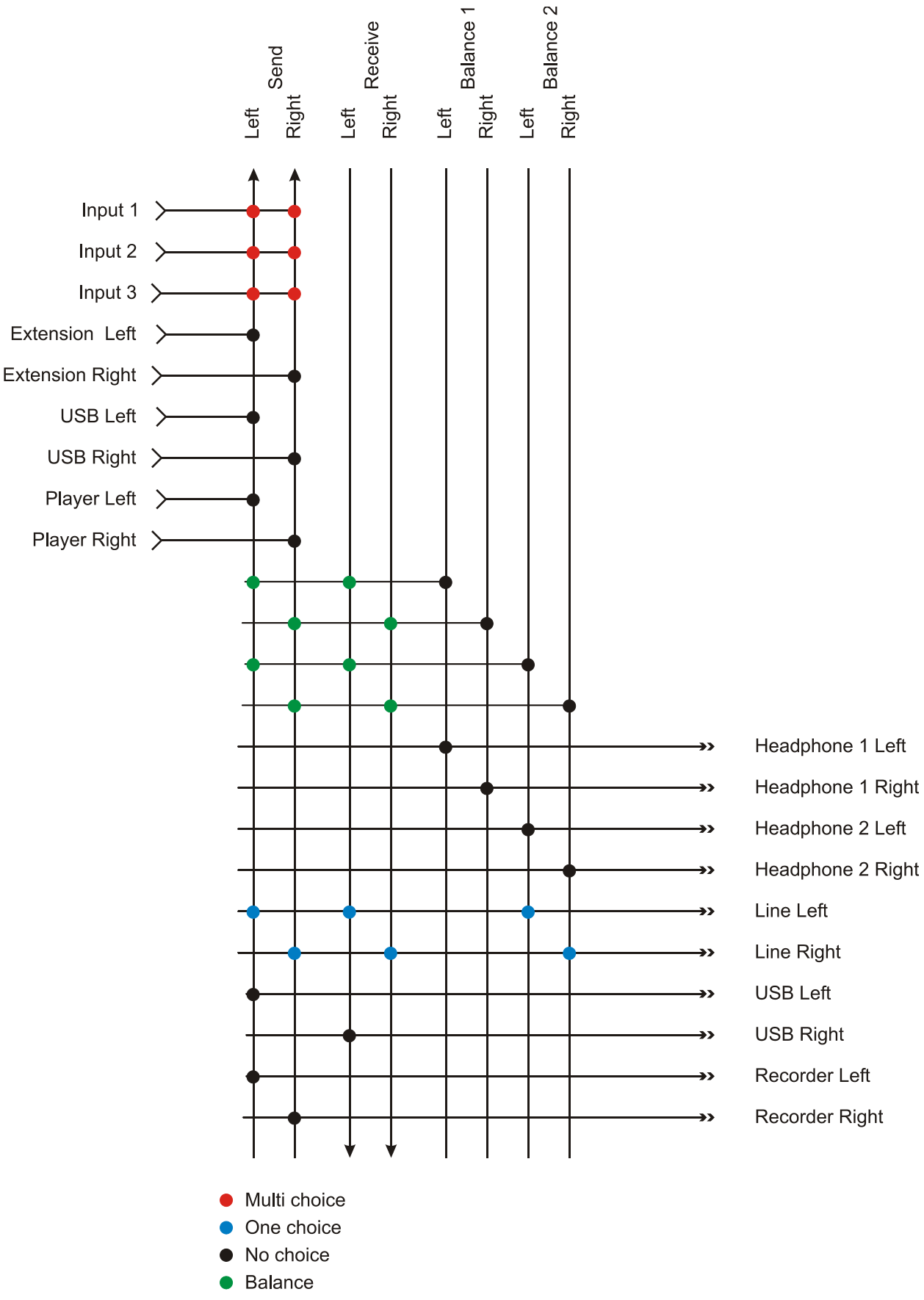
MONO ROUTING : PROGRAM



MONO ROUTING : PROGRAM & COORDINATION



MONO ROUTING : PROGRAM & INTERNATIONAL SOUND



STEREO ROUTING

2.4. Auxiliary functions

The main function of the SCOOPY+ is the transmission of one or two main audio programmes, but it also provides auxiliary functions for transmitting data or additional signals, inside the same stream (or, more generally, the same session).

These features are only compatible with AETA products, because they are not inside the scope of independent standards.

The availability of these functions depends on the coding algorithms, and on the transmission network. The following tables show these capabilities for the various networks.

ⓘ *No auxiliary function is available for mobile voice transmission. Besides, for PSTN/POTS only "relay transmission" (4 loops) is possible.*

Codec	Relays Isolated	Data (bauds) 300	1200	2400	4800	9600
G711						
CELP						
G722-SRT						
G722-H221						
G722-H242	X	X				
TDAC						
4SB ADPCM	X	X				
MPEG L2						
MPEG L2/J52						
AAC-LC						
HE-AAC						
HE-AAC v2						

X = exclusive (only one function at a time)

Table 4 – Auxiliary functions: over ISDN

Codec	Relays Isolated	Data (bauds) 300	1200	2400	4800	9600
G711						
CELP						
G722						
G722-H221						
G722-H242						
TDAC						
4SB ADPCM						
MPEG L2						
MPEG L2/J52						
AAC-LC						
HE-AAC						
HE-AAC v2						

Table 5 – Auxiliary functions: over IP networks (wired or mobile)

2.4.1. Transmission of relays

You can transmit 2 states (Contacts) through a data channel during an establish link.
These contacts can mange relay or "ON AIR" Lamp on the remote side.

Note: In ISDN & POTS, the feature should be enabled on both sides. Some algorithms don't support it in ISDN.

Enter the menu by pressing OK, set Tools / Misc / Aux. Functions/ Relays to ON.

During the connection, with

and

you can change the state of the remote relays. From the main window, you can see the state of them by pressing the joystick to the right.

LED Info 1 and 2 show you the state of the remote contacts, ie local "relays".

2.4.2. Use isolated relays

To access this feature, you should connect an external USB relay box on your Scoopy+ (Contact your dealer for the compatibly).When this function is activated, you can set feature for each of them.

- Inputs
  - Info 1/2: the codec transmits to the remote unit the status of the isolated current loops.
  - Redial: Redial the last phone number with the same configuration (same as twice)
  - Release : Hang-up the call (same as)
- Relays
  - Info 1 / 2: the remote codec transmits its input state. The unit then opens or closes relay contacts according to the transmitted status.
  - Line state: The unit closes relay contacts when the connection is established
  - Sync. State: The unit closes relay contacts when audio are decoding correctly

For transmission over IP, this feature is always available whatever the coding algorithm. For the other networks, availability depends on the algorithm: see above tables.

With G722 or 4SB ADPCM, relay transmission cannot be used at the same time as another auxiliary function (see tables).

A typical application is the transmission of "on air" signals; a contact closure can be used for instance to light up an indicator or switch on other equipment.

2.4.3. Data channel

This function is available in ISDN transmission mode.

In ISDN mode, a bi-directional data channel can be transmitted along with the compressed audio signals, by reserving a fraction of the transmitted bit rate.

The data are transparently transmitted end-to-end; hardware signalling is not available. Availability depends on the coding algorithm, as well as the max possible baud rate: see above tables. With G722 or 4SB ADPCM, this data channel cannot be used at the same time as another auxiliary function (see tables).

On Scoopy+, the data channel is only used for remote control with AARC protocol from the data channel interface of the remote codec.

Note : contact your distributor for more details

2.4.4. Coordination channel

This function is available only with Mobile option. It enables the transmission of an auxiliary audio channel (or coordination channel) in voice mode through mobile network. This transmission can be established in parallel of any over transmission.

Each audio input can be assign on this auxiliary channel through its menu configuration.

2.5. Supervision and control interface

The control and supervision of the equipment (configuration, communication management, status monitoring), is carried out either "locally" thanks to a keyboard, an alphanumeric display, LED indicators, or using various remote control interfaces.

2.5.1. "Local" control

For local management, the front panel includes a large keypad, a graphic OLED, and various LED indicators for essential status information.

2.5.2. Embedded html server: "web pages"

SCOOPY+ provides html pages that enable full control using a web browser, via port 80 (default port for the HTTP protocol). See on chapter 4.11 the detailed operation mode.

This control mode can be used from any computer regardless of its OS (or a mobile device with a web browser), and the embedded pages are compatible with all current browsers. No software installation is needed on the control device.

2.5.3. Additional Ethernet/IP interface

It is possible to get a second Ethernet/IP interface by plugging a USB/Ethernet adapter[1] on the USB socket. This interface can be used as a remote control port (html pages via port 80, or "command line" mode via port 6000), as an alternative to the normal integrated interface. *It should not be used for other functions (audio over IP).*

2.5.4. Configuration and dialling memories

To ease the operation, it is possible to store configuration memories, called "profiles". These belong to three categories:

- "Remote profiles", including the parameters for calling a given destination: dial numbers, coding algorithm, etc.
- "Local profiles" which memorise the network access characteristics. Recalling a local profile is a quick way to recover the configuration needed for connecting on a given line/network.
- "Snapshots" which memorise all the settings for the audio interfaces.

These various profiles can be used locally and also through the web pages, and they can be imported/exported from/to a computer.

[1] Reference : Logilink UA0144, or other device shown on our web site support pages.

2.5.5. Test functions

For maintenance purposes, some test loops can be activated. The following drawing schematically shows these test loops:

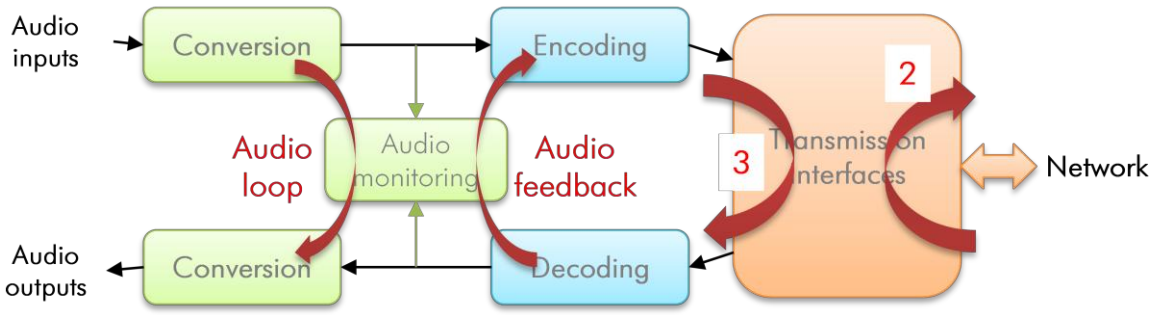


Figure 5 – Test loops

- "Audio loop": uncompressed audio data are looped from the input of the encoder to the input of the output conversion functional block. This loop redirects the audio input to the audio outputs;
- "Loop 3", or "Codec" loop: compressed audio data are looped just before the network interface;
- Loop 2", or "Network" loop: this loop sends the received data back to the network; for the remote codec, the effect is the same as a loop 3 when the transmission works correctly;
- "Audio feedback" loop (audio output to audio input); this allows the codec to send back to the remote codec the signal it receives, after decoding and re-encoding.

3. Operation

3.1. General principles – Control means

The equipment control and supervision (configuration, status monitoring) is possible either in “Local” mode (front panel keypad and display, status indicators), or “Remote control” mode, thanks to an Ethernet interface.

As a general rule, the configuration parameters are saved in non-volatile memory, and restored when the unit is powered on.

Local mode operation is described in detail in chapter 4 (Detailed operating mode).

Remote control operation using a computer and a web browser, thanks to the embedded HTML server, is detailed in chapter

.

The SCOOPY+ can be remote controlled by third-party codec management software and systems. Please consult us for more information on the available offer in this field.

3.2. Physical description of the equipment

The SCOOPY+ codec is housed in an ABS chassis.

3.2.1. Front panel

All the elements needed for local controls are located on the front panel (see picture on page 31 below). This panel can be roughly divided in many parts:

One can find several LED indicators, an LCD and the main navigation keys. Another part can find dialling keys and call management keys. Lastly, one can find audio control elements.

On/Off switch and standby

First, on the left is located the

on/off key, and just besides the ON indicator, keep the key pressed for at least 3 seconds to switch on the unit. When it is operating, keep the key pressed for at least 3 seconds to switch it off.

Status LED indicators

The LEDs have the following meaning:

Marking	Color	Function
DC/Charge	*Green*/*Red*	Off when unit is off and hasn't DC power; *Green* when a DC power is pluged; *Red* when internal batteries are in charge
DEC	*Green*/*Red*	Off when unit is idle; *Green* when the decoder is synchronised on "Line 1"; *Red* if interface is active/connected but not synchronised or in error
LIM **(3 times)**	*Green*/*Red*	Off when limiter on associated input is off; *Green* when the limiter feature is enable by menu; *Red* if the limiter is working
ON **(3times)**	*Red*	On when Microphone input is enable and routed

Figure 6 – SCOOPY+ front panel

1 - Headphone 1 : Local/Cue (external ring) and volume adjustment (central knob)

2 - Input 1 volume control

3 – Input 2 volume control

4 - Headphone 2 : Local / external ring) and volume adjustment (central potentiometer)

5 – Joystick

6- OLED Screen

7 – keypad

"INFO" indicators

These LEDs show the state of the *received* information when the auxiliary function "Relay transmission" is active:

Marking	Color	Function
INFO 1	*Amber*	Shows the state of the "relay contact" n°1
INFO 2	*Amber*	Shows the state of the "relay contact" n°2

Audio management keys

Key	Function
One "Mute" key for each input	Mute the input, ON led shows the state (RED : input enable) The initial mute state can be defined by menu

LCD and navigation keys

Besides the backlit graphic LCD one can find the keys for navigating through the menus:

Key	Function
OK	Confirm a selection or enter data. From the base screen: switch to the root menu.
Esc	Escape to upper menu level; From the base screen: switch to the root menu.
Navigation key : Joystick	Keys used to move the cursor or browse through menu options. The ▶ can be used to enter a sub-menu (like the OK key). When entering data/numbers, the ◀ erases the character on the left of the cursor. A Confirmation (Same as Ok key) is available on the Joystick

Call management and dialling keys

Key	Function
"Hang up"	Release an established link. *(except LL links, permanent by nature)*
"Unhook"	Start a link or accept an incoming call. *(except LL links, permanent by nature)*
Dialling keys **0 .. 9** ***.** **#⇑**	Keys for entering numbers or URI for the destination to call. These keys are also used to enter texts such as profile names, etc. Repeatedly press a key to get the characters other than the main figure. Keys **1** and "***.**" also allow entering special additional characters (not all marked on the keypad). The "**#⇑**" key is used for switching between numbers, lower case letters or capital letters. An indicator on the screen recalls the current type of characters.

Figure 7 Recording keyboard

Function keys

The assignment of these **F1** and **F2** keys is programmable.

3.2.2. Rear panel

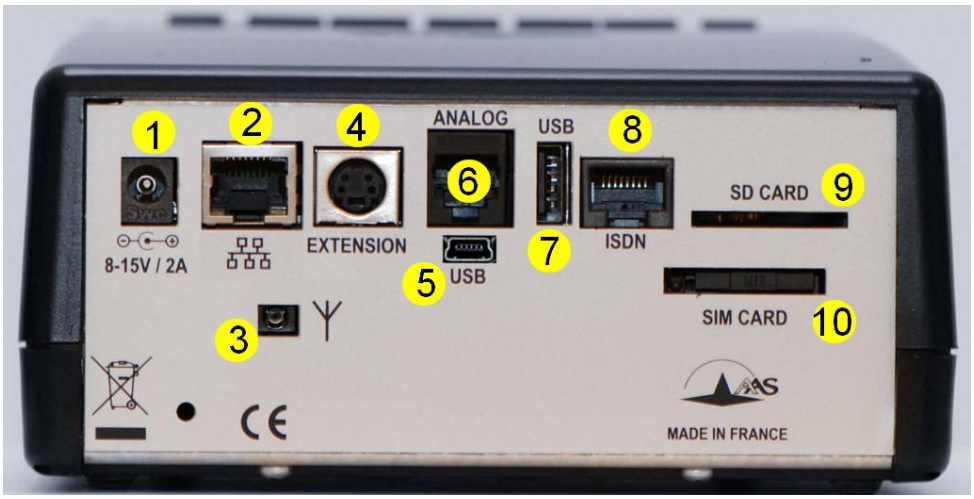


Figure 8 – Rear panel

1. External DC 2.1mm Jack 2. Ethernet RJ45 Socket 3. External antenna connector for Integrated GSM module 4. Audio device extension connector, compatible with Mixy 5. USB (Mini-B) to connect to PC	6. POTS RJ11 Jack 7. USB interface to connect with any USB peripheral 8. ISDN RJ45 Socket 9. SD card holder 10. Removable holder for SIM card

Ethernet interface [2]

This socket is a 100BaseT/10BaseT port, used for audio transmission over IP and/or for remote controlling the unit. This RJ45 socket is devised for a normal "straight" cable to an Ethernet hub or switch. The two integrated LEDs show the presence and activity of the network (green LED) and the interface mode: half-duplex (yellow LED off) or full-duplex (yellow LED on).

The configuration of the interface is described in 3.4, Initial setup of the Ethernet interface.

ISDN socket[8]

This RJ45 socket allows the connection to the ISDN, for the product versions which include this capability. The wiring is for connecting an S0 (BRI) ISDN line using a standard RJ45 cable.

Only this socket has to be used when only one line is needed (links using one or two B channels), which includes the operation in double ISDN codec mode.

POTS socket (marked "Analog") [6]

For the units equipped with the PSTN/POTS option, this socket is used to connect the POTS line. You must plug here a cable from the POTS wall socket[1] and terminated with a RJ11 plug.

[1] Whose type depends on the country and the building wiring

Drawer for SIM card [10]

In the units fitted with the mobile network access option, this drawer houses the SIM card enabling the access to the network and services.

ⓘ *The SIM card must be inserted while the unit is off (or in standby).*

To remove the drawer and the card, push the small button shown by the little arrow besides the drawer (*see picture below*).

To set the SIM card in place, first insert it in the cavity inside the drawer. Make sure to have the cut corner on the right place, with the SIM chip visible. Then insert the whole assembly into the slot, with the SIM card chip facing down. Check that the drawer is well in its guides before pushing it completely in place.

ⓘ *If you have to use a smaller format µSIM card, you can use a SIM/µSIM adapter (available on request from AETA): first set the µSIM inside this adapter, afterwards use this assembly like a regular SIM card.*

Antenna sockets [3]

On the products fitted with the mobile network access option, these HIROSE MS-151NB sockets allow to connect an external antenna (a multiband antenna is included with the mobile network option). In this case the internal antenna is automatically disconnected.

The antennas must cover the band(s) used for the operator and network services. Refer to the operator in doubt. The provided antenna covers the 900 MHz, 1800 MHz, 1900 MHz, 2100 MHz bands. It is compatible with almost all the 2G and 3G/3G+ networks in Europe.

USB A socket [7]

This "host" socket allows the connection of a peripheral device, e.g. a mobile access USB module or "key" in order to access mobile IP transmission.

Refer to chapter 2.1.4 about this function.

As another use, an Ethernet adapter can be plugged for providing a secondary Ethernet/IP remote control interface; *see chapter 2.5.3 for this topic*.

USB Mini-B socket [5]

This "device" socket allows the connection to a PC in order to use the Scoopy+ has a sound card.

Refer to chapter 2.1.4 about this function.

"Extension" socket [4]

This socket allows connecting AETA AUDIO SYTEMS Mixy product for having more audio inputs.

SD card [9]

SDCard is used to record mixed signal for later diffusion.

3.3. Installation and set up

3.3.1. Mounting and connections

To operate the codec, the minimum necessary connections to set up are (see details in the rear panel description):

- Power supply (Batteries and/or DC);
- Audio inputs and outputs (XLR sockets);
- Network interface: depending on the networks used, Ethernet interface, ISDN line(s), POTS line, or antenna(s) for mobile network access[1];

The pin out of the connectors is indicated in chapter 6.1: Characteristics of interfaces.

3.3.2. Initial set up

Before the first use, the equipment must be configured according to the desired operation mode: audio input/output format, local conditions (network interface parameters…). Then to set up links you must select the coding type and parameters.

For using the configuration menus, a password may have to be entered. After factory setting or after a complete configuration erasure, the password is "blank" (no password) and the access is unrestricted. Afterwards, if needed, a password can be programmed by the user and the restricted access mode can be activated.

For more details about the codec configuration, see chapter 4 - Detailed operating mode. The setup for the Ethernet interface is described in 3.4 below (Initial setup of the Ethernet interface).

[1] In such case, a SIM card should be set as well in order to enable the mobile services.

3.4. Initial setup of the Ethernet interface

The SCOOPY+ includes a 100BaseT / 10BaseT Ethernet interface, and the audio transmission can take place over an IP network through this interface. In addition, it is always possible to use the Ethernet interface to access the embedded html server or for remote control the unit via a TCP/IP connection (TCP port 6000).

An initial set up is needed for using one of these features of the Ethernet interface. For setting into operation, first connect the Ethernet interface to the network, using CAT5 wiring.

- Connection to 10BaseT or 100BaseT interfaces are both suitable, as the SCOOPY+ automatically switches to the adequate 10 Mbit/s or 100 Mbit/s mode.
- "Straightforward" patch cables should be used for a connection to a hub or a switch. Conversely, a "crossed" cable might be needed for special configurations (e.g. a test connection to a PC).

As a very first step, the Ethernet interface must be assigned an IP address, and related parameters. This phase is very simple when a DHCP server is available in the network. The menu to use is reached by **Setup** / **Network** / **Ethernet Parameter**.

3.4.1. DHCP server available

This is the simplest case, because the server will allocate a suitable IP address and give the unit the right settings. Select "**DHCP**" in the menu (**Setup** / **Network** / **Ethernet Parameter / Mode**). The unit will then automatically find the DHCP server and automatically set the parameters. You can read the IP address (allocated to the unit by the DHCP server) in the "About" menu (**Tools / About**).

Note that, as an additional advantage with DHCP, you do not need to change this setting later, even if you move the codec to another network, as long as it is still connected to a DHCP server.

3.4.2. "Static" IP configuration

When there is no DHCP server, you have to enter the settings manually. The IP address must be "available", i.e. not already assigned to other equipment. Ask support from the network administrator(s) as needed.

First select the manual mode, menu **Setup** / **Network** / **Ethernet Parameter** / **Mode** / **Manual.** Then, still in the menu **Setup** / **Network** / **Ethernet Parameter,** you must enter the following parameters:

Parameter	Notes
Local IP	Must be unique on the network
Subnet Mask	A typical value is 255.255.255.0
Gateway	
DNS Server 1	Domain Name Server (main)
DNS Server 2	Domain Name Server (secondary)

All addresses are in "dot-decimal" format, such as e.g.: 192.168.0.12, 10.0.54.123.

ⓘ *Note: in contrast to the configuration with DHCP, the "static" setting has to be reviewed each time you move the unit to a new physical site/network, as the previous IP addressing is probably not valid for the new location.*

3.4.3. Checking the IP configuration

The above configuration is kept in the unit's memory, and reloaded at each start.

To check the setting, you can read the IP address in the "About" menu (**Tools / About / Local IP**).

You can then also check that the unit is seen on the network and at the right address: from a computer connected to the same network, enter (in the command mode, or console mode depending on the OS) "ping *ipaddr*", where *ipaddr* is the IP address of the SCOOPY+.

If the response is positive, then you can proceed with the rest.

3.4.4. Optional configuration of the "link" Ethernet layer

With factory settings, the Ethernet interface is set for an auto negotiation of the "link" mode: speed (10 or 100 Mbit/s) and half-duplex or full-duplex.

ⓘ *This setting is suitable for almost all situations and usually you don't need to change it.*

However, in the rare cases where this is needed, it is possible to set the desired mode manually: use the menu **Setup** / **Network** / **Ethernet Parameter** / **Link Mode** . The available choices are:

- **Auto-negotiation** (*standard setting*)
- **100BaseT, full-duplex**
- **100BaseT, half-duplex**
- **10BaseT, full-duplex**
- **10BaseT, half-duplex**

Whenever this setting is needed, it must be done before the IP configuration described above.

3.4.5. Configuration of the secondary Ethernet interface

In case an external USB/Ethernet adapter is used to provide an additional interface (see 2.5.3), an initial setup is also required for this interface. The procedure is similar to that for the main integrated Ethernet interface:

- First plug the adapter on the USB Interface.
- Connect the Ethernet interface to the network, using CAT5 wiring. *Normally the adapter's LED should blink depending on the network activity*. The additional interface is designated "**Ethernet 1**" in the unit's menus.
- Apply the same steps as described above (3.4.1 to 3.4.4), the settings are in this case accessed via the sub-menu **Config** / **Network** / **Ethernet 1 Parameter**.
- To check the IP address and status of the interface, go to the menu **Config** / **Network** / **Ethernet 1 Parameter / Network Config.**

3.5. Managing links

3.5.1. Setting up and releasing links

Generally speaking, once the codec is set up and the transmission interface(s) to be used is(are) configured, it is possible to manage audio links with remote devices.

A transmission links/sessions have to be set, which can be done in two ways:

- "Outgoing call" launched towards a remote device: the procedure is to select a transmission interface, a coding configuration, "dial" the destination to call and then set the call to the remote unit. The detailed operating mode is described further, for the various transmission media, in 4.5 (Setting up a link).
- "Incoming call" received from a remote device: on receiving a call on one of the connected and active interfaces, the codec switches to this interface and processes the call.

In a similar way, ending/releasing a link is either initiated by the remote unit (remote release), or by the operator of the SCOOPY+ (local release).

3.5.2. Auto-redial feature

The IP mode and the ISDN modes are "dial up modes", where a link can be set up and released at will. When it is necessary to hold the link on permanently, outgoing calls may be backed up by using the auto-redial function (for each codec in the double ISDN mode). When it is active and the codec is the initiator of the link, the codec automatically tries to re-set the link in case of an initial failure, or if an established link is dropped for another reason than a local release (i.e. hanging up by the user). The redial capability applies in the following situations:

- If the initial call fails for any reason (e.g. called party is busy); the codec then redials and retries to establish the link.
- The codec can also redial if the link is already established and the link is lost, for any reason else than "local release" (e.g. the remote unit mistakenly dropped the line).
- After a power failure, after rebooting the codec will automatically redial and set-up the link back.

ⓘ *Note that, while "auto redial" is active, an established link can be definitively stopped only by releasing the line on the calling codec side. Otherwise, every time the called party will hang up, the calling codec will redial and reset the link.*

It is possible to program the time period that the unit will wait before redialling after a failed trial, and it is also possible to program the maximum number of times the codec will redial before giving up.

The activation of this function and the configuration of its parameters can be found in the "Auto Redial" sub-menu (**Tools** / **Misc** / **Auto Redial**). In dual codec mode, the function can be activated separately for each codec.

3.6. First level maintenance

3.6.1. Using the test loops

The test loops accessible from the "**Tools** / **Maintenance** / **Tests** " menu can help improve the analysis of a problem:

- In order to check if the audio part functions correctly, use the "Audio" loop and check if the audio is OK at the output.
- To check if the coding part functions correctly, activate "Loop 3" [1] and check if the alarm disappears (and the decoding indicators come back to normal), and if the audio is present at the output.
- "Loop 2"[2] sends back to the remote codec the compressed data received from the network (see the description of test loops in 2.5.5, Test functions). In this way, it is possible to test the integrity of the transmitted data and/or check that the remote codec works properly.
- The decoder out to encoder in loop ("Audio feedback" loop) can be used for overall functional check, and also for aligning the overall chain.

Note: not all test loops are available for all the network interfaces.

[1] Note: this loop is not available for all transmission configurations

[2] This loop is not available for all transmission configurations

4. Detailed operating mode – User interface

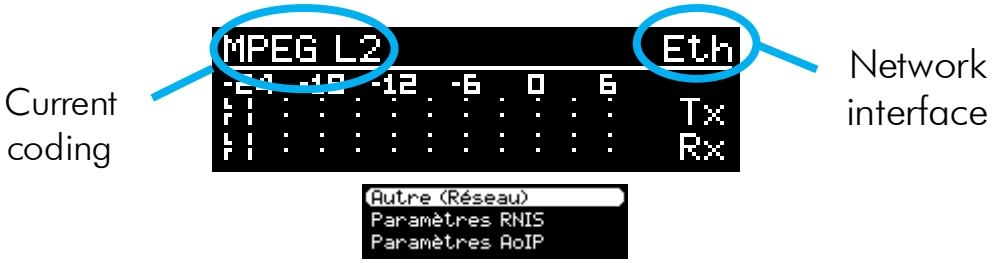
In local mode, the unit is operated thanks to a keypad and display on the front panel. The display is an OLED graphic display.

Operating from the keyboard can be protected by a password (numeric sequence, that can be programmed or erased by the user. In such case, the operation is in a "restricted" mode, without direct access to settings or dialling. Only the profiles menu is accessible, for loading a profile or call a remote profile (*see later in 4.7,* Restricted operation mode *for more about the restricted mode*).

4.1. Equipment start-up

Press the ⏻ key for at least 3 seconds to start up the unit.

During start-up, the unit displays temporary messages. This initialisation lasts around 30 seconds. Then the main menu is displayed.



On this "base screen", transmission and reception audio levels are permanently monitored for the two audio channels.

At this time, the unit can be operated and receive calls. It is also possible, using the menus, to change settings and/or set up calls.

ⓘ *If the unit has been set in "restricted access" mode, for recovering full access to all menus type directly the password (numeric sequence), and the device unlocks as soon as the last figure of the password is pressed.*

See further in 4.7 how to use and manage this restricted access.

4.2. Principles for the navigation

From the base screen, if pressing the Esc key the display switches to the highest level menu, called "root menu"[1].

Pressing Esc again brings back the base screen.

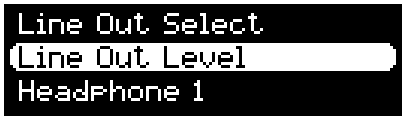
The LED indicators and the main keys surrounding the LCD are described above in 3.2.1, Front panel

The unit proposes a tree structure of menus, and the Joystick is used to navigate through the menus. The OK key or the Joystick contact is used to confirm choices or enter values for parameter, while the Esc key allows to come back up to the upper menu level. Repeatedly pressing this key brings you up to the root menu, and the base screen.

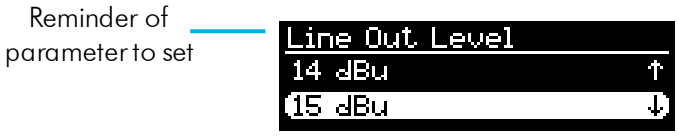
From the root menu, you can enter one of the three main menus by using the arrows and pressing OK to enter the highlighted menu.

From these main menus, the screen is organized in the following way:

The sub-menus have the following aspect:



At the end of a "branch", lists of choices appear in this way:



In all cases, the principles are as follows:

- Use the Joystick to browse sub-menus or choices.
- Enter the highlighted sub-menu by pressing OK or the ▶ key.
- Move to upper level with the Esc key or the ◀ key
- Validate a choice or entered value using OK
- Give up a selection or action by pressing Esc (repeatedly as needed)

ⓘ *After some time without pressing any key (about 30 seconds), the display comes back to the base screen.*

[1] You can also press the OK key

4.3. Dialing and text input keypad

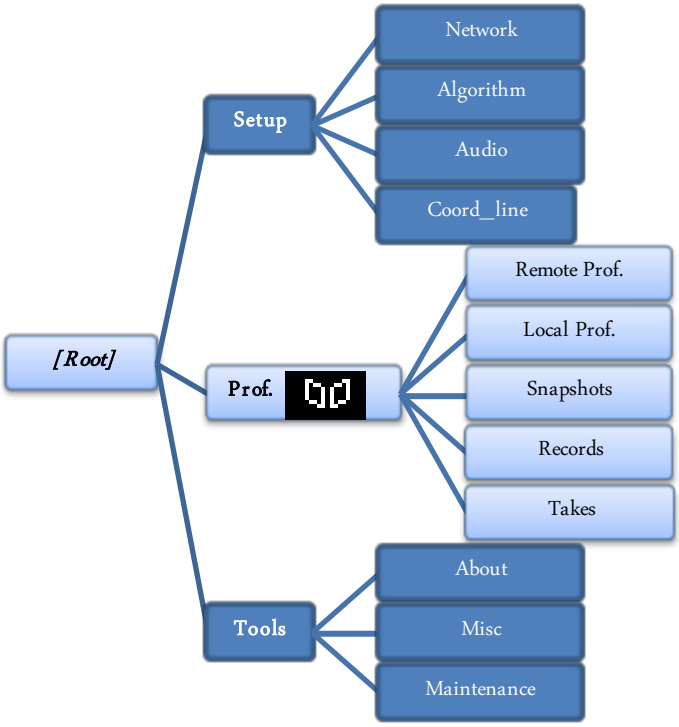
The keypad is used for entering numbers and/or texts. It is used in a similar way to a mobile phone:

- The keypad works in "numeric" only mode, or in "alphanumeric" mode (where both numbers and letters can be entered).
- In numeric mode, only the numbers are used, and the "*****" key (as a separation between ISDN address and sub-address, or as a dot in a numeric IP address).
- The "alphanumeric" mode allows text entry. A letter can be entered by pressing repeatedly a key; for example, a "B" is entered by hitting the "**2**" key two times (sequence A, B).
- Use the **#⇑** key to switch between "numeric", "alphanumeric (lower case)" and "alphanumeric (capital)" modes.
- The "**0**" and "**1**" keys provide access to various special characters which are not all printed on the keypad: "**@**", "**:**", "**-**", "**_**"…

4.4. Description of the menus

The root menu shows the following three main menus:

- **Setup** configuration of the codec
- (profiles): access to configuration memories or "profiles"
- **Tools** Maintenance and miscellaneous functions, access to status information



The « **Setup** » menu leads to the following sub-menus:

- **Network** selection and configuration of the network interface and parameters
- **Algorithm** selection and configuration of the coding algorithm
- **Audio** configuration of the audio interface and parameters
- **Coord_Line** Establish a coordination line through a mobile connection

The menu for profiles leads to the following sub-menus:

- **Remote Prof.** selection and edition of remote profiles
- **Local Prof.** selection and edition of local profiles
- **Snapshots** selection and edition of snapshots (memories for audio parameters)
- **Records** Manage recorded files
- **Takes** Manage takes (extraction of part of recorded files)

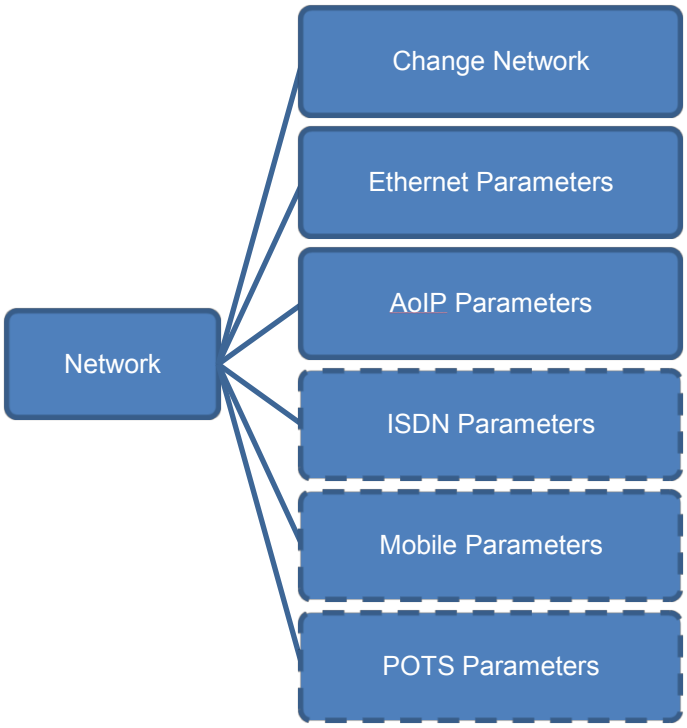
ⓘ *This menu is the only branch of the menu tree that is accessible even in restricted mode.*

The « **Tools** » menu is itself divided in sub-menus:

- **About** display information on addressing and firmware version
- **Misc** miscellaneous functions
- **Maintenance** maintenance and test functions

The following diagrams show the various sub-menus and the available parameters.

4.4.1. Network sub-menu



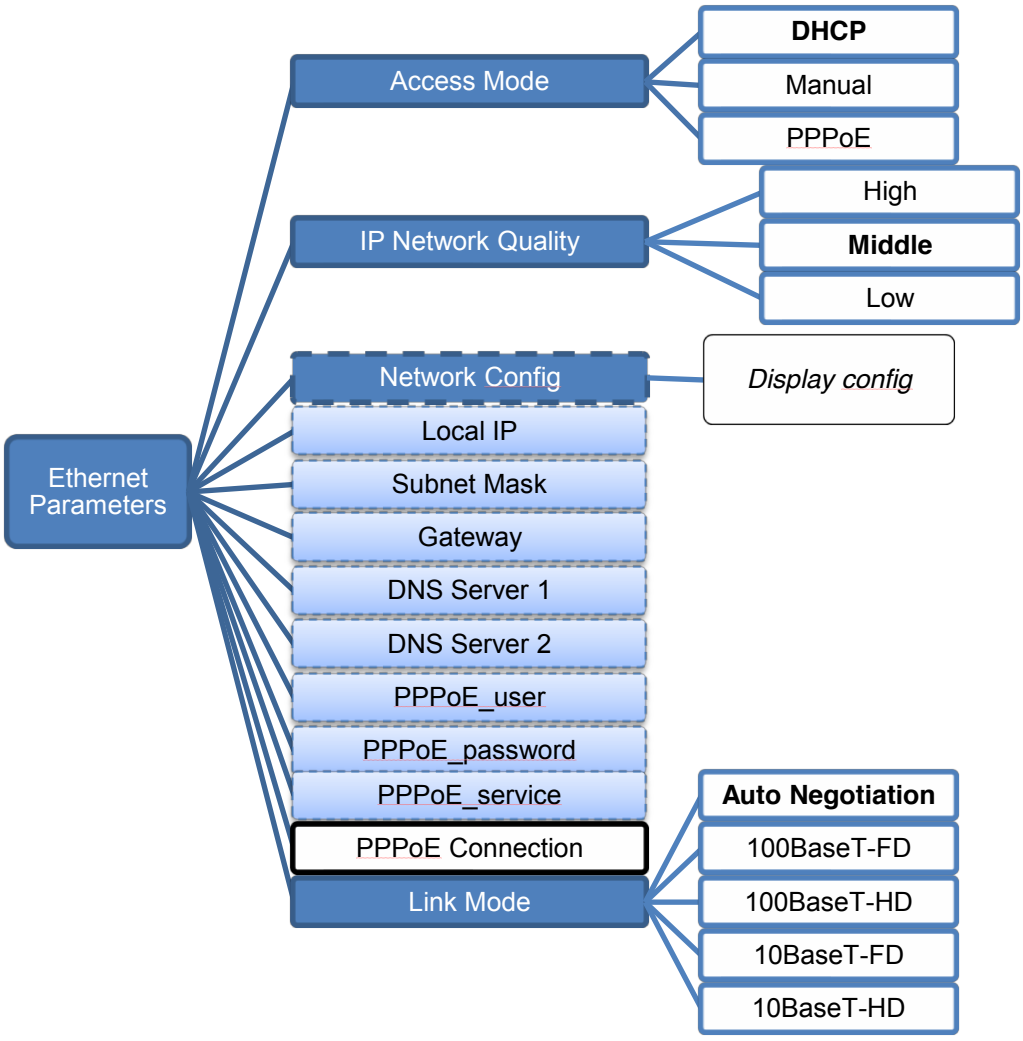
The **Change Network** sub-menu shows a selection among the available network interfaces. This is how you select the "default" interface, which is used for an outgoing call. This menu must be used before setting a call (except if the default interface is already the desired one). The current interface is recalled on the right side of the screen:

The other sub-menus provide access to the settings for each existing interface. This is why some interfaces (*with a dashed frame outline on the picture above*) may not be visible in the sub-menu depending on the unit version, if it does not include such interfaces.

ⓘ *If you have plugged a 3G or LTE "mobile key" on the USB interface, you can also see this module in the list, and you can select it just like another network interface, designated as "Mobile 1 (Ext.)". If more than one mobile device is available (example: presence of an integrated module and also an "external" device connected on the USB socket), then each is identified in a unique way.*

ⓘ *In a similar way, if an Ethernet adapter is plugged on the USB Interface, you will see in the list a new interface, named "Ethernet 1". The corresponding sub-menu has the same structure and settings as the sub-menu "**Ethernet Parameters**".*

Ethernet Parameters



Notes:

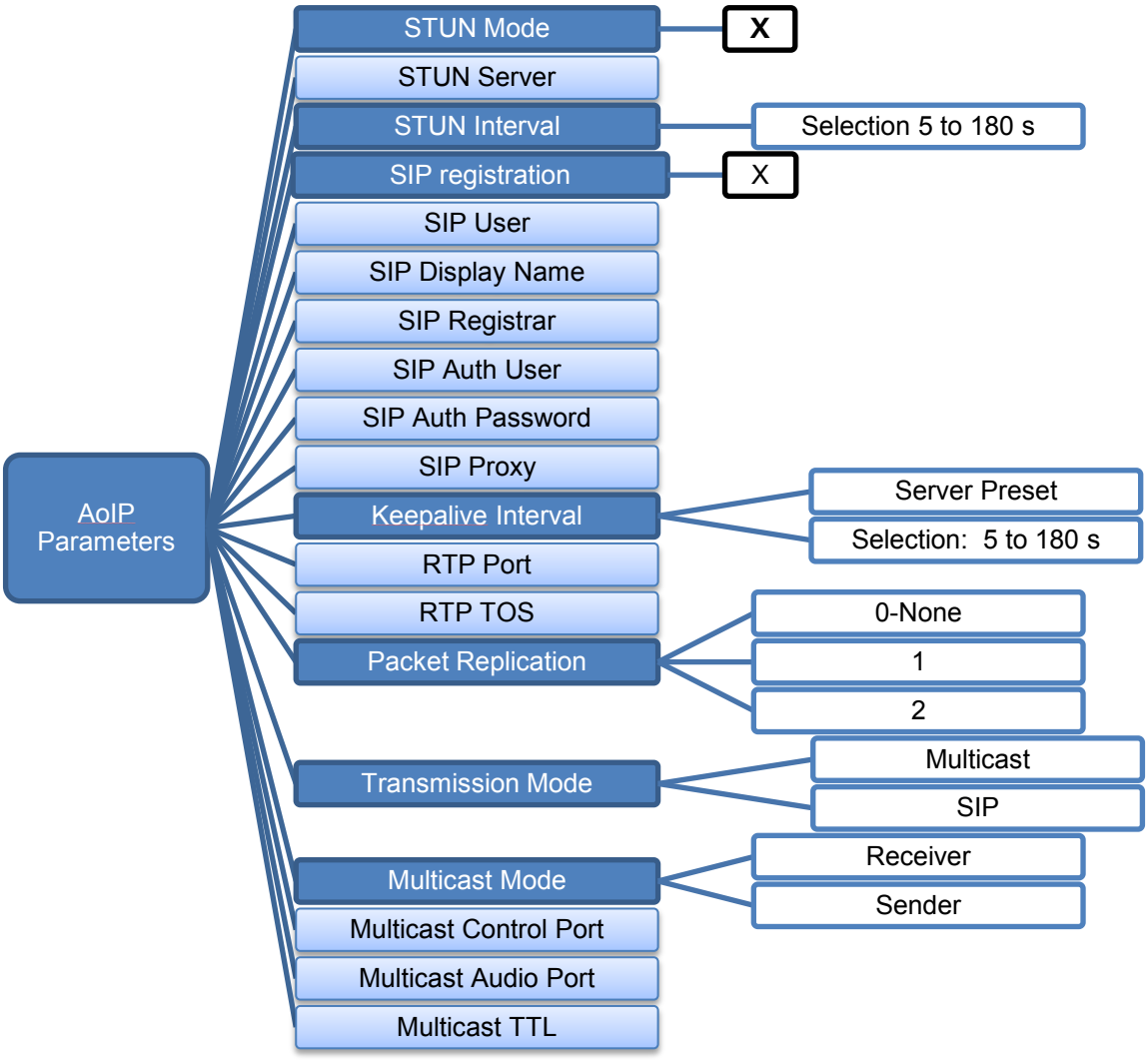
- **Access Mode**: the current setting is recalled on the right side. The selection is operative immediately after selecting an addressing mode.
- **Network Quality**: select here the expected quality of the transmission via the Ethernet interface. The unit switches to internal settings suitable for this quality level.
  For instance, if you select "**Low**", the codec sets a large size for the reception buffer, in order to stand a higher jitter. As a counterpart, the latency will be higher as well.
  Conversely, if you select "**High**", the buffer is smaller and the latency is lower, but the system is more vulnerable to possible jitter (fluctuations of the network transmission latency).
- **Network Config**: only appears when DHCP is active, in which case this sub-menu displays the addressing parameters that the SCOOPY+ got from the server:

  Using the arrows, you can browse the various settings: local IP, sub-network mask, etc. *Exit this display with the* Esc *key.*
- If DHCP is not used, several menus allow you to enter the needed settings, such as **Local IP**, **Gateway**, etc. (*lighter background on the picture above*)

- **Link Mode** is related to the physical Ethernet interface; the default setting "**Auto-negotiation**" is suitable for almost all situations.

AoIP Parameter

This sub-menu groups the settings related to the Audio over IP (AoIP) transmission. It is relevant for the Ethernet interface but also for using a mobile access network, be it with the integrated module or an external module via USB.



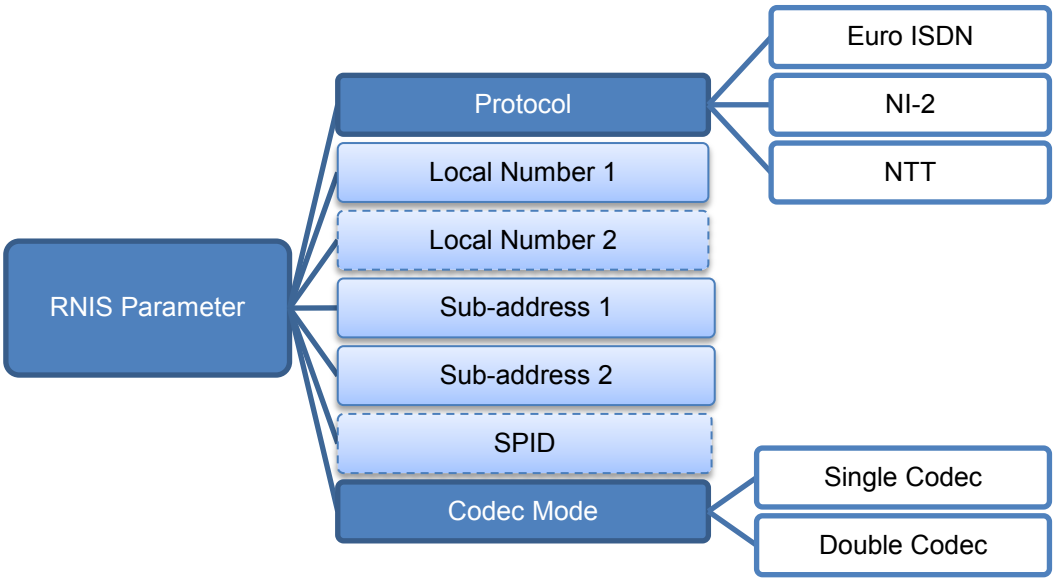
Notes:

- **STUN Mode**: enable or disable STUN. *In this way you can disable STUN without erasing the STUN server address.*
- **STUN Server**: enter the address of the STUN server, either in numeric form or with a symbolic name. This kind of server allows SCOOPY+ to discover its public IP address when it accesses the Internet through a router with NAT. *You can use AETA's server:* ***stun.aeta-audio.com.***
- **STUN Interval**: interval between two queries to the server; usually you can keep the default setting of 15 s.

- **SIP Registration**: enable or disable the registration on a SIP server. The following menus SIP xxx are used to enter the parameters of the account on this server, when using such service. These parameters should be provided by the administrator of the SIP server. The **SIP User** field usually corresponds to a number that a remote party can call to initiate a link with the codec when it is registered on the server.
  With this **SIP Registration** sub-menu, *you can disable the registration and hence stop using a SIP server, without having to erase account parameters; they stay ready for later use.*
- **Keepalive Interval**: sets the interval between two re-registrations on the SIP server. This setting can be used to make the registration renewal more frequent, but the server always imposes a maximal period; for this reason the default setting "**Server Preset**" is recommended except for specific cases.
- **SIP port**: the default port for SIP signalling is *5060* (UDP), you can enter a different value here.
- **RTP Port**: the default port for RTP/SIP audio streams is 5004 (UDP), but you can enter a custom value here if desired.
- **RTP TOS**: if the transmission network supports DiffServ to manage the media priority, you can set here the type of service which is assigned to the stream transmitted by the codec.
- **Packet Replication**: (see also 2.1.1 and page 8 regarding this feature) you can select here the setting for packet replication:
  0 => standard mode without replication (default setting)
  1 => packet replication without interleaving
  2 => packet replication with interleaving: the second packet is delayed; robustness is improved but latency is increased.
- **Transmission Mode**: select here between SIP (unicast) or using multicast . *The following sub-menus exclusively apply to the multicast mode. Besides, the SIP settings are not relevant for this multicast transmission mode.*
- **Multicast Mode**: if the selected mode is Multicast, the transmission is unidirectional. Select with this menu if the codec must be *sender* of a multicast stream, or rather *receiver* of such stream.
- **Multicast Control Port**: this port number (UDP) is used for sending control packets accompanying the multicast stream. Default value is 6000.
- **Multicast Audio Port**: this port number (UDP) is used for transporting the multicast audio stream. Default value is 6001.
  *Note: this setting is not related to the **RTP Port** setting valid for RTP/SIP.*
- **Multicast TTL**: TTL value applicable to the RTP multicast audio stream. Default: 254.

ISDN Parameter

This sub-menu is not available on versions with no active ISDN interface.

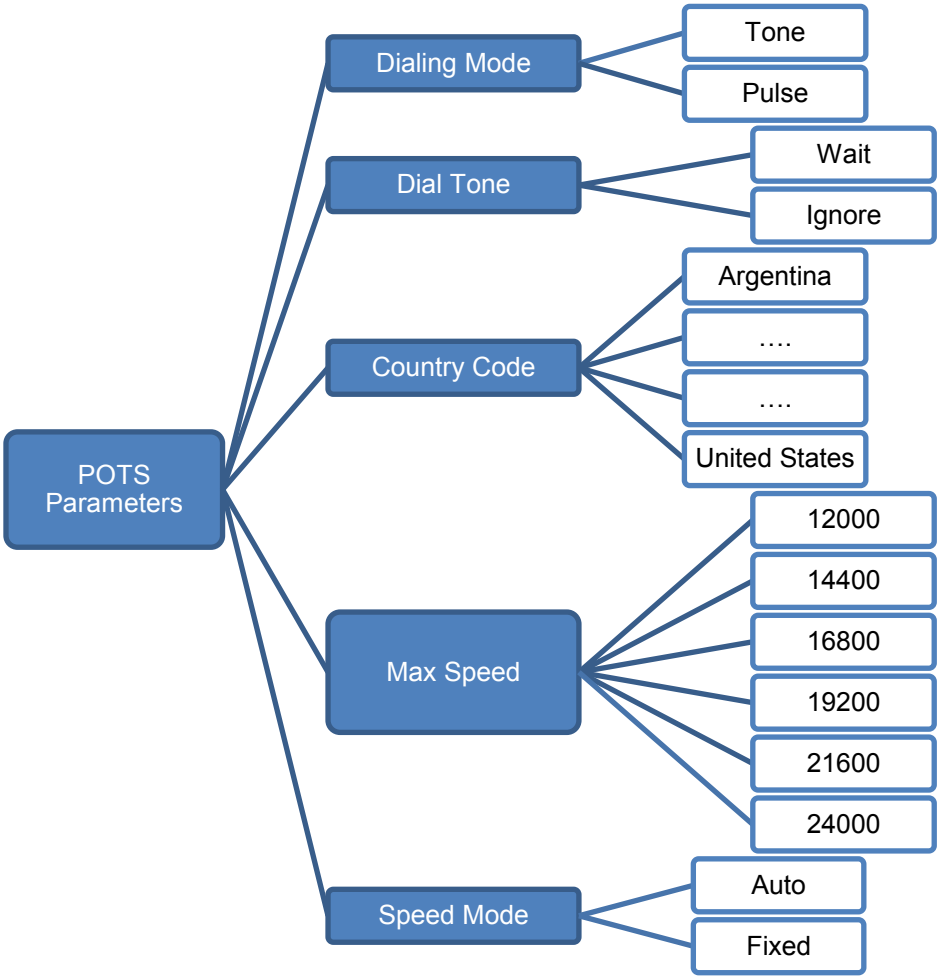


Notes:

- **Protocol**: NI-1 is used in North America, NTT is the protocol of the Japanese operator NTT, and Euro ISDN (or ETSI) can be used in the majority of the other countries (*see more details on page 9, Network protocol*).
- **Local Number** and **Sub-address**: these settings are irrelevant for the NI-1 protocol. *Their meaning and use is detailed further, page 61, in 4.5.1,* Setting up a link in ISDN mode.
- **SPID**: this setting only applies to the NI-1 protocol, and is not presented for the other protocols. See page 61, in *4.5.1,* Setting up a link in ISDN mode, more about this setting.
- **Codec Mode**: selection of either the normal "single codec" mode or the "Double codec" mode. In this latter case, it may be important to set a distinctive local number + sub-address combination for each of the two first channels. This is detailed *page 63, in 4.5.2,* Setting up links in double ISDN codec.

POTS Parameter

This sub-menu is only visible on units with the POTS option.



Notes:

- **Dialing Mode**: tones are used by default (DTMF), but for specific cases (old PBXs, etc.) it is possible to select pulse dialing.
- **Dial Tone**: choose "**Wait**" so that the codec waits for the dial tone before dialing the numbers. The dial tone may be specific with some particular equipment or networks, and may not be detected by the SCOOPY+; in such case, disable this function ("**Ignore**", and the codec will dial the numbers just after picking up the line. *This latter choice also works correctly with most modern PBXs.*
- **Country Code**: this choice will set the internal modem for the characteristics applicable in the selected country. The full list is not shown above as it is quite large! For a country not found in the list, usually it is possible to select a country with similar standards (such information may be found from local specialists).
- **Max Speed**: in nominal use, the POTS codec runs at 24000 bit/s, but it can automatically adapt according to the line quality, and change the bit rate if required at the beginning and during the link. With this setting, you can constrain the bit rate to a lower value. Reason for this: on a line featuring a variable quality[1] you can avoid undesirable changes during the communication by "limiting" the modem to a speed that you know to be stable. The speed may nonetheless go down further if the quality decreases too much.

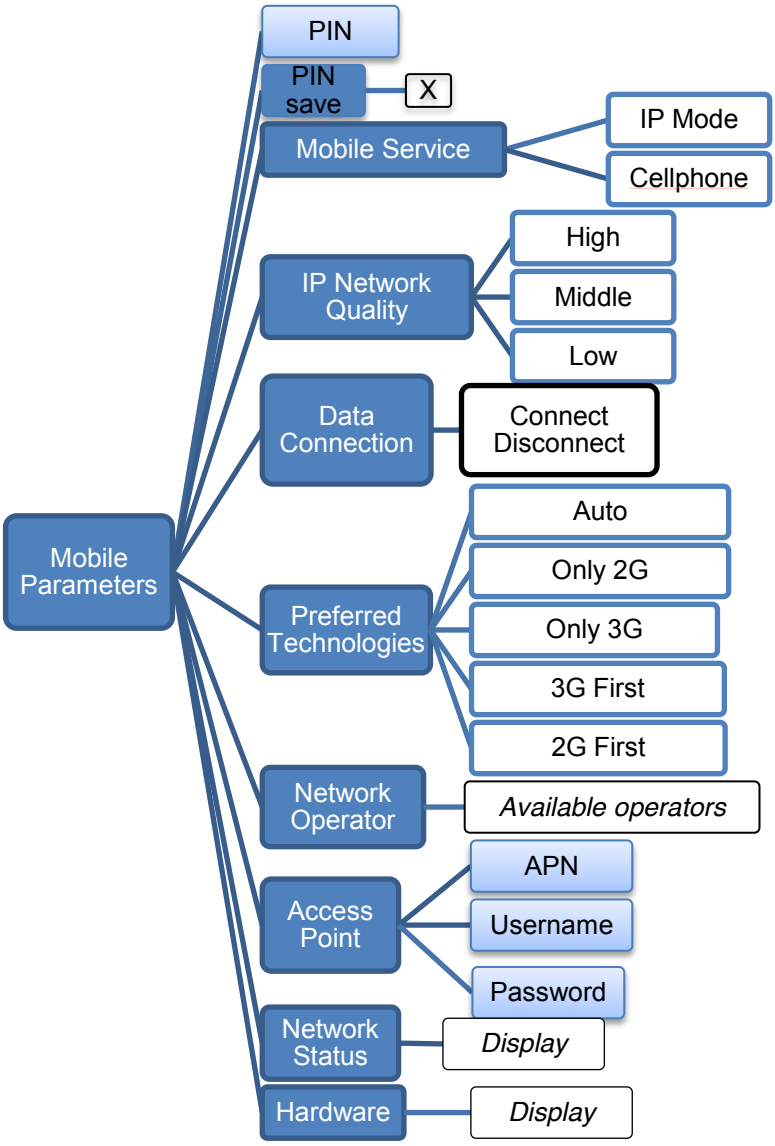
[1] This can happen for example when the line suffers a strong crosstalk from neighboring lines.

- **Speed Mode**: for a line/transmission with variable quality, and/or to avoid repeated changes of the bit rate, with this setting you can force a fixed bit rate. In such case, the "Max Speed" above setting becomes a fixed value. Be aware you must be sure that transmission conditions will not require a lower speed; otherwise you incur a synchronization loss in such eventuality.

Mobile Parameter

This sub-menu is only visible on units with the "wireless" option.

If more than one mobile access module is available (example: integrated module and "mobile key" on the USB socket), then one such sub-menu is presented for each module.



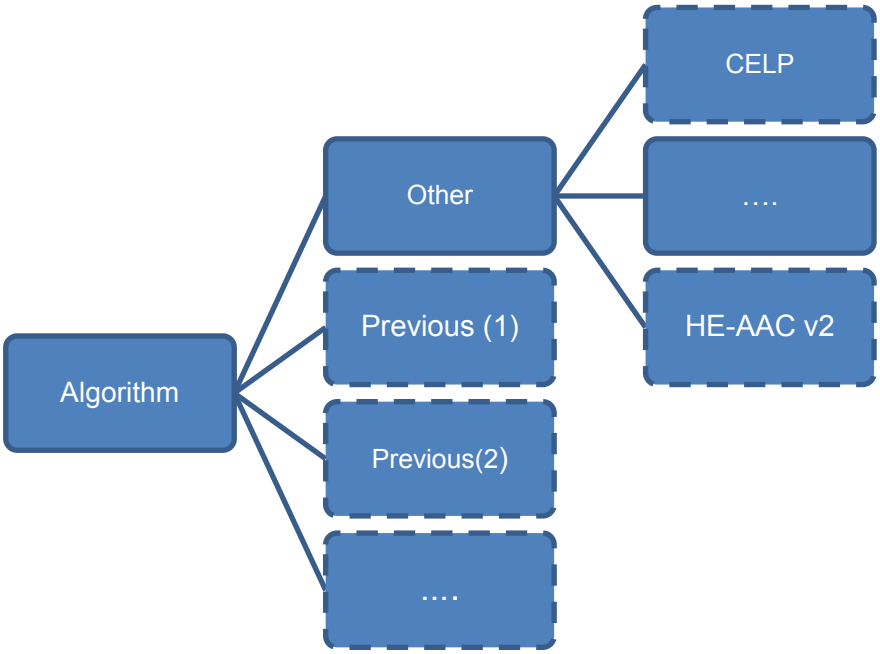
Notes:

- **PIN**: enter the PIN code allowing to use the SIM Card and access the network. Entering this code is the first step necessary to operate the mobile access, except if the code for the card is empty or disabled. You can save your PIN code inside the scoopy+
- **Mobile Service**: select the desired transmission mode: "**IP Mode**" for transmission of data packets (IP protocol), also called "PS" mode (for Packet Switched), or "**Cellphone**" for the telephone mode, which can also be wide band "HD Voice" when all conditions are met (*see above in 2.1.4, Mobile voice mode – HD Voice*).
  Note: the "cellphone" mode is not available for a "mobile key" device on the USB interface, because such devices only support IP data transmission.

- **IP Network Quality**: select here the expected quality of the transmission via the mobile interface in IP mode. This is the same type of setting as described (see above in 45, Ethernet Parameters) for the Ethernet interface, but here applying specifically to the mobile interface. These separate settings allow to take in account differences in quality and link conditions between the two interface types.
- **Data Connection**: applies to the IP mode; use this to activate the data connection, or disconnect it if it is active. This is similar in function to connecting an Ethernet cable on a wired network: by activating the connection the unit is linked to the Internet and it can send and receive IP data. For instance it can receive an incoming call. It can also initiate an outgoing call.
- **Preferred Technologie**: select the network technology. "**Auto**" is the normal choice, which provides operation on 3G/3G+ as long as the conditions allow it, but fallback to GSM if 3G is not available or the radio link is too poor. It is also possible to force either the 3G/3G+ modes ("**Only UTMS**" selection), or the GSM/2G mode ("**Only GSM**" choice).
- **Network Operator**: normally the unit automatically selects a suitable network. If you want to force a specific operator (which can happen when using the unit abroad), this menu presents the list of available network so you can manually select one. *But not all networks are actually allowed, this depends on the policy of the SIM card's provider!*
- **Access Point**: this sub-menu allows to enter the necessary parameters for data connections (IP mode): APN, user name and password. The essential setting is the **APN** (Access Point Name), mandatory for accessing the mobile IP network. SCOOPY+ automatically sets the APN for many operators, but you may have to enter this yourself, or change the default setting. The other parameters can usually stay empty; otherwise they must be provided by the network operator, along with the SIM card and the APN.
- **Network Status**: information on the mobile network (for the current connection): network/operator identification, technology (GSM, EDGE, UMTS, HSDPA, HSPA, etc.), radio signal strength. *Note: bars on the base screen also show the mobile reception level)*
- **Hardware**: information on the internal mobile access module (type, firmware, IMEI) and about the SIM card (IMSI, subscriber number).

4.4.2. Algorithm sub-menu

This sub-menu allows to select the desired configuration for an outgoing call, using the current transmission interface.

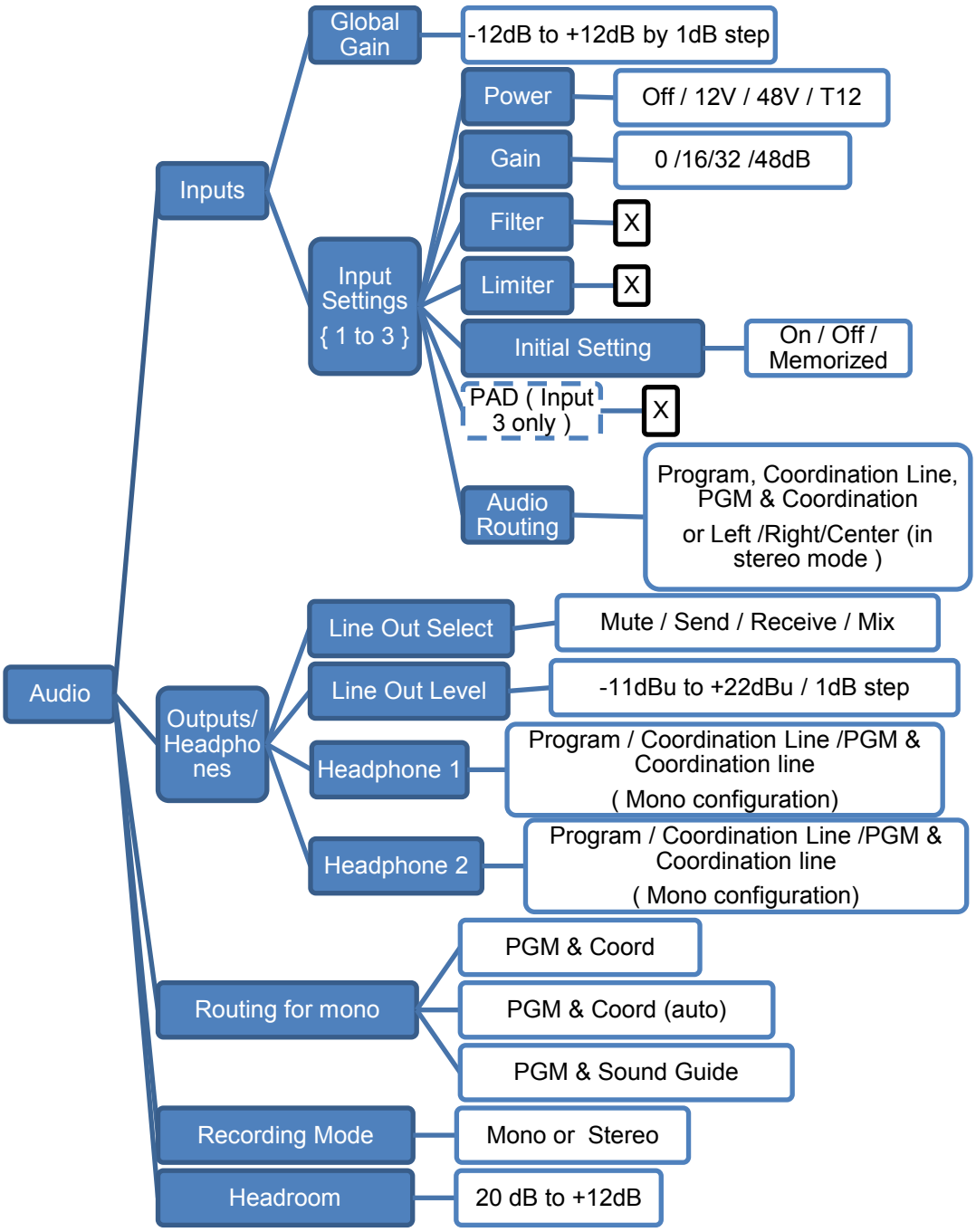


It is possible to quickly recall the last used configurations ("Previous" above), scrolling the list and selecting one of these recent settings.

Otherwise, selecting "**Other**" you get the list of the coding algorithms available for the current network interface and the installed options. Choices that are not compatible with the network interface or not installed are not displayed.

After selecting the coding algorithm, for some of these you must select additional parameters, such as: channel mode (for stereo-capable coding), sampling frequency and bit rate (for MPEG L2 and AAC coding), protection mode (only for MPEG J52 and CELP)

4.4.3. Audio sub-menu



Notes:

- **Global gain**: set the gain on the transmit mixed
- **Input Settings:** Settings for each microphone/Line input
  - **Power :** configuration of the microphone power. Be care full, enable T12 only if it is needed by the microphone
  - **Gain :** Set the microphone gain
  - **Filter :** Enable the analogue high pass filter at 50Hz 18dB/Oct to remove low noise frequency
  - **Limiter:** Enable the analogue limiter (set @-6dBFs)
  - **Initial state :** Mute state on power on
  - **PAD :** A 20dB attenuation is available only on the input 3 to support high input.

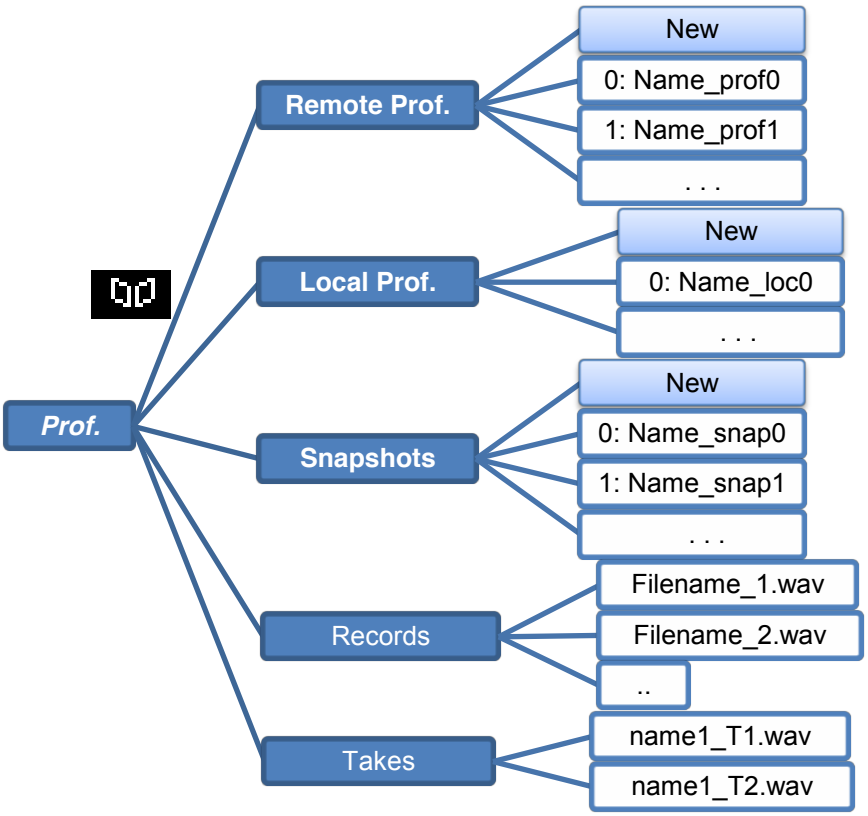
- **Audio routing:** Setting depend of the routing configuration(mono /stereo & coordination channel/international sound)

- **Output/Headphones**
  - **Line Out select:** You can selected sources from the Send, Receive or the Mix of the headphone 2.
  - **Line Out Level:** gain setting for the outputs. The parameter is the maximum level (absolute level expressed in dBu) of the interface. For an input this is the clipping level, which when applied at the input produces a transmitted signal at 0 dBFS. For an output this is the maximum output level, produced when receiving a signal at maximum level, i.e. 0 dBFS.
  - **Headphone x :** Sources selection depending of routing configuration(see also 2.3.2 Audio routing)
- **Routing for mono**: This menu isn't accessible in stereo configuration. Here is the selection of the international sound and coordination channel. See 2.3.2 Audio routing for more details
- **Recording mode** : To select mono or stereo record format.
- **Headroom**: this setting defines the reference level for the program level display. This display is relative to a "0 dBr" reference which can be set here. The "Headroom" parameter is the available headroom for a signal displayed as 0 dBr, i.e. the difference between the reference and the 0 dBFS maximum level. For instance, a headroom of 12 dB means that the reference is set at -12 dBFS.

On the display, the "OVL" clipping indicator shows that the absolute maximum level is over -3 dBFS.

4.4.4. Profiles menu

This is the only menu available in restricted access mode.



Notes :

- **Profiles**

"Profiles" are configuration memories that can be recorded and later recalled at will (*see their description in 2.5.4 above*). For each profile category, a sub-menu provides the list of profiles already recorded in the SCOOPY+ memory, each with its index and name.

On the top of the list, the "**New**" choice allows you to create a new profile:

- Enter a name for this new profile, then OK. For a "local profile" or a "snapshot", the current settings (respectively network settings or audio settings) are recorded into the new profile with the name you just entered.
- For a remote profile, first you can elect to include or not the coding parameters: select "**Current**" (include the current codec settings) or "**None**" (do not include an algorithm with this profile).
- Then you can enter the dial number (or IP address, or SIP URI), then OK. *You can also leave this field empty; in this case the profile is used for memorizing only coding parameters, rather than a remote number.*
- Lastly, for ISDN and a setting exceeding 64 kbit/s, you must enter an additional number (or more).

To use the profiles, just scroll the list, select one of the recorded profiles, and press OK: a choice is presented:

- **Load** this profile, recalling the memorized parameters;

- **Call** the remote destination in the profile (choice available only for a remote profile): in this case the codec recalls the memorized parameters, and then sets the call to the memorized number(s);
- **Delete** the profile (not available in restricted access mode);
- **Update** the profile (for a remote profile): you can edit an existing profile, going through the same steps as for creating a new remote profile. *This choice is not available in restricted access mode.*

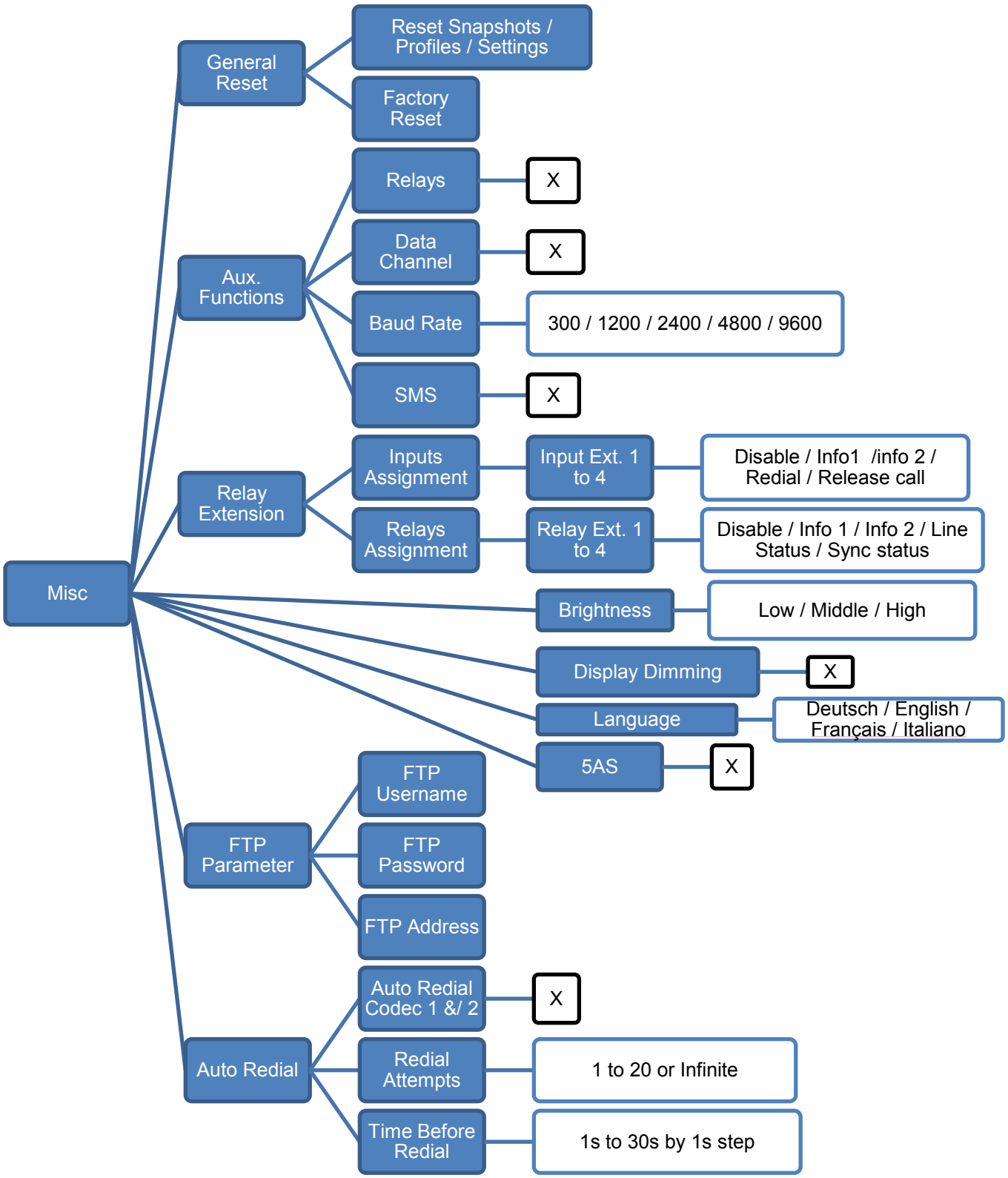
- **Records**
  - **Records :** Here you will find your recorded file. To use the files, just scroll the list, select one , and press OK: a choice is presented :
    - Play : to play back in the mixed inputs
    - Rename : to change the filename
    - Delete : To remove this file from the SD card
    - Send : To send this file to a FTP server (See Tools / Misc / TFP Parameter)
  - **Takes :** Here you find prepared records for transmission. To use the files, just scroll the list, select one , and press OK: a choice is presented :
    - Play : to play back in the mixed inputs
    - Rename : to change the filename
    - Delete : To remove this file from the SD card
    - Export : To generate a new file from selected part of the record file.

4.4.5. Tools / About sub-menu

This sub-menu provides a list of data about the identification and status of the unit. Scroll the list using the arrows and access all the information:

- **Local IP**: IP address of the active IP interface (Ethernet or mobile depending on the current one).
- **Public IP**: public IP address, when SCOOPY+ accesses the Internet through a NAT router. This address is usually detected thanks to a STUN server (configured via the "AoIP parameters" menu, see in 4.4.1).
- **Version**: SCOOPY+ firmware version global identification. *Along with the unit serial number (shown on the label stuck on the rear panel), this is part of the essential information to provide whenever you contact AETA for support or maintenance issues.*
- **EIM Version**: version number for the audio via IP module.
- **SW Build**: date of firmware generation.
- **MAC Address**: of the main (integrated) Ethernet interface.
- **Ethernet Setup**: current setting of the main Ethernet interface (10 or 100 Mbit/s, half/full duplex…)
- **SIP-Registrar**: address or domain name of the SIP server, if one is configured
- **SIP Status**: shows whether the codec is registered on the SIP server.
- **Battery level** : show battery voltage

4.4.6. Misc sub-menu



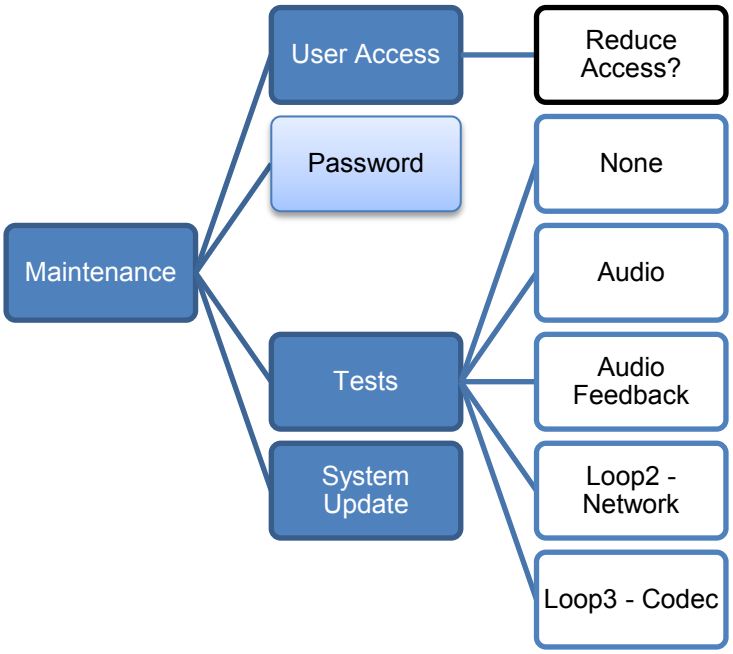
Notes:

- **General Reset**: you can select to delete only the snapshots or the profiles, or conversely to reset all parameters but keep these memories ("**Reset Settings**" choice). For a complete erasure with the restoration of all default settings, select "**Factory Reset**". *Caution: this cannot be cancelled!*
- **Aux. Functions**: this sub-menu allows to enable the auxiliary functions ((details on these functions can be found in 2.4, Auxiliary functions), and select the baud rate for the data channel. Besides, it also allows to enable or disable the displaying of the received SMS on the LCD (active by default).
- **Relay Extension** : Selection of feature available on each contacts and relays
- **Brightness**: select the preferred front panel display brightness
- **Display Dimming**: After 20minutes of inactivity, the display switch the brightness to low
- **Language**: select the preferred language, and the menus are immediately updated.
- **5AS**: by default the 5AS system for ISDN (cf. page 9, 5A System®) is active. If required, it can be disabled using this sub-menu.
- **FTP** : You can configure the access of your FTP server to send your record files
- **Auto Redial**: this sub-menu deals with the auto redial feature, whose operation is described in 3.5.2 above. For the double ISDN codec mode, auto redialling can be enabled separately for each codec.
- **Redial Attempts**: you can adjust from 1 to 20 the number of attempts to reset the link in case of a drop, or even "**Infinite**" for redialing indefinitely.
- **Time before redial**: (in seconds) adjust between 1 and 30 seconds, after a link loss, the waiting time before redialing. It is often worth leaving such a pause, so that the cause of the link loss is possibly cured, or the remote unit gets back to its normal state, etc.

4.4.7. Maintenance sub-menu

On one hand this sub-menu allows managing the restricted access mode, which is specifically described further in 4.7, Restricted operation mode.

On the other hand it allows activating test loops (as described in 2.5.5, Test functions).



- **User Access**

For setting up the restricted access, first you must set a password (that will be needed to later unlock the device), and for this use the "**Password**" sub-menu.

To activate the restricted mode, select the "**User Access**" sub-menu, and confirm by selecting "**Yes**". Then you must enter the password to lock the unit (this in order to make sure you know this password!). The unit immediately enters the restricted mode and will stay in this mode (even after a power cycle or reboot) until it is unlocked using the password.

ⓘ *If you lose the password: it is still possible to apply a complete factory reset; this will also clear the password and restore full access. The various methods are recalled in chapter 4.8 further, Clearing all settings.*

- **Tests :** See 2.5.5 test function for more details
- **System update:** Make an update from the SD CARD

4.5. Setting up a link

4.5.1. Setting up a link in ISDN mode

Preliminary settings

The network interface must be configured depending on the local ISDN line that is used, thanks to the "ISDN Parameter" sub-menu (**Setup** / **Network** / **ISDN Parameter**)

Protocol

The default setting is "Euro ISDN", also known as ETSI protocol. Change this setting if another protocol is needed in your location.

Local address

In some cases, it may be necessary to set the local address (or local ISDN number) of the line, and/or it is possible to assign a sub-address to the codec.

The local number allows "multiple subscriber numbering" or MSN. This number is usually the number remote equipment must dial to call your equipment. Configuring this number in the equipment is not mandatory if the equipment is directly connected to the public network. On the other hand, if the equipment is connected to a PABX, the number(s) are often required. The PABX may also impose a unique number for each B channel within the same BRI interface. In such a case, refer to the characteristics and configuration of the PABX.

ⓘ *Proper configuration of the local numbers is essential, and many problems in setting up links originate from mistakes or misunderstandings regarding this configuration. In doubt, leave this number blank! This is usually appropriate for public lines.*

Sub-address SA

This number differentiates several terminals connected to the same ISDN bus, which are allocated the same call number(s). Thus it can be useful in case other devices are connected with the SCOOPY+ on the same line.

Whenever a sub-address is set, the unit will only accept incoming calls specifically directed to this sub-address.

ⓘ *Most often, the best setting is to leave this blank, except in double codec mode where this setting is useful (see further the processing of calls in double codec mode).*

SPID

This "Service Profile Identifier" is only relevant for the NI-1 protocol (used by some operators in North America). This number identifies the services of the ISDN line, and usually includes the subscriber number or part of it. This number is mandatory to operate the line, and must be provided by the network operator.

Calling an ISDN number

To set the link, first set –if not already done- the unit in ISDN mode (menu **Setup** / **Network** / **Change Network** / **ISDN**). Then select the desired encoding format (**Setup** / **Algorithm**).

ⓘ *You can get a fast setting by simply selecting one of the "recently used" configurations displayed; otherwise select "**Other**" for a new coding configuration.*

Come back if needed to the base screen using Esc. Then enter the ISDN number of the destination and press the ✆ key.

If a sub-address is needed, right after the number enter the "*" character and the sub-address (4 digits max.). The number then has the form nnnn*ssss, e.g. 0912345678*32 (then press).

When the selected encoder needs two B channels, the unit asks for additional numbers. If the same number is suitable, just press the "green phone" button without re-entering a number.

An error message is displayed in case of a failure of the link establishment, or on a remote release. *To come back to the base screen, press* Esc *or* .

Receiving calls

When the 5A System is active, receiving calls is simple. When a call is received, the codec automatically "unhooks" and recognises the coding algorithm and protocol used, and finally sets the link automatically. On the receiving side, the unit will "follow" the calling unit.

When the 5A System is not active, you should first configure the codec for the desired coding algorithm and configuration. When a call is received, the unit will synchronise with the calling device, but the link will usually fail if the calling party has used another coding configuration than expected. *However, if J52 is used by both parties, the link will succeed even without 5AS active.*

ⓘ *Important notice: the unit can receive and accept incoming ISDN calls even when its current network interface is not ISDN, except if it is busy with an established link.*

Releasing/Hanging up

In any case, to stop a link, press the key.

ⓘ *Press a second time to confirm! (this is for safety, in order to avoid releasing by mistake)*

Quick redialing

Once a number (or more for ISDN links with more than one B channel) has been called, it is easy to recall it without having to type it again: press the key, then you can scroll through the "history" (last dialled numbers) using the arrows. Press again the key when the desired number is displayed. This is especially efficient for quickly redialling the previous number.

4.5.2. Setting up links in double ISDN codec mode

The double codec mode is specific, as the unit behaves as two independent G711/G722 mono codecs, designated "Codec 1" and "Codec 2". Each codec sets or releases links separately on the B channel it is assigned, and uses the audio input and output it is assigned: input and output A (or "left") for Codec 1, and input and output B (or "right") for Codec 2.

Preliminary settings

The network interface must be configured depending on the local ISDN line that is used, just like in the normal ISDN mode (see above 4.5.1). However, in this mode, configuring a sub-address on each B channel (hence each codec) can be very useful. Indeed this is a means, when receiving calls, to differentiate the two codecs (which often answer the same subscriber number as they are on the same ISDN line). *But be aware these addresses cannot be dialed from a remote analog telephone.*

Besides, make sure to select the double codec mode: … / **ISDN Parameter** / **Codec Mode** / **Double Codec.**

Calling an ISDN number

To set the link, first set –if not already done- the unit in ISDN mode (menu **Setup** / **Network** / **Change Network** / **ISDN**).

Note that the display (base screen) has changed to show the respective status of the two codecs.

Then select for the relevant codec the desired encoding format (**Setup** / **Algorithm**): first select the codec (Codec 1 or Codec 2), then select the algorithm.

Come back if needed to the base screen using Esc. Then enter the ISDN number of the destination (followed by * + sub-address if needed) and press the key: you should then select the line/codec by pressing the F1 or F2 key depending on the codec you want to use.

An error message is displayed in case of a failure of the link establishment, or on a remote release. *To come back to the base screen, press* Esc *or* .

Receiving calls

Receiving a call works in the same way as in te single codec mode, except for the following:

- If the codecs are differentiated with exclusive subscriber numbers, a remote call can be sent specifically to one of the two codecs.
- You can have the same capability using the sub-addresses (*but this cannot be used by a remote analog phone*).
- Only G711 and G722 algorithms are supported by the codecs ion this mode.
- If the call is not specifically addressing one of the two codecs, the first available one (priority to codec 1) will pick up the call. Example: if codec 1 is busy, then codec 2 will pick up the call.

Releasing/Hanging up

To stop a link, press the key. You must then select which line/codec you want to release, using key F1 or F2. Press a second time to confirm.

Quick redialing

Once a number has been called with a codec, it is easy to recall it without having to type it again: press the key, then you can scroll through the "history" of calls using the arrows. Press key **F1** or **F2** depending on the desired codec.

4.5.3. Setting up a link in IP mode via Ethernet

A link is set up in a similar way as an ISDN link. The difference is mainly that instead of the telephone number, we use either an IP address, or a SIP URI (Uniform Resource Identifier).

Preliminary settings

If a SIP server is used, the SCOOPY+ must be registered on this server. Use the **AoIP Parameter** menu (*see above on page 46*) for entering the SIP account data and activate the SIP registration). *Check that the codec is successfully registered, using the **Tools** / **About** sub-menu: **SIP Status** information.*

Besides, for links over the Internet, most often the link passes through a NAT router which "hides" to the codec the characteristics of the public access: public IP address and RTP ports actually used. This can make the SIP transactions fail, or get a non-optimal route through the network. To avoid this a STUN server is often used; the corresponding settings are also found in the **AoIP Parameter** sub-menu.

*Check that the public address discovery has succeeded: **Tools** / **About** sub-menu: **Public IP** information (its mere presence shows the discovery has succeeded).*

ⓘ *You can find in annex 7.3 (Some methods to deal with NAT routers and firewalls) some additional information on the use of STUN.*

Directly call an IP address

This is the most basic way of setting the link. It is suitable only if:

- The other unit is "directly" reachable, i.e. there is no NAT Router or firewall blocking the connectivity. The simplest case is when both units are on the same LAN or private network.
- The IP address of the other unit is known!

To set the link, first set the unit in IP mode (**Setup** / **Network** / **Change Network** / **Ethernet**), and set the desired encoding format (**SETUP** / **Algorithm**, etc.).

ⓘ *Note that you can get a fast setting by simply selecting one of the "recently used" configurations displayed; otherwise select "**Other**" for a new configuration.*

Then enter the IP address and press the key.

ⓘ *When operating in this way, it is recommended to disable the SIP registration (**SIP Registration** parameter in the **AoIP Parameter** sub-menu, see on page 46).*

Calling via a SIP server

This is the technique when both units are registered on a SIP proxy server. In this case, each unit is identified by its SIP URI, in the form username@sipservername, like an email address. There is no need to know any IP address (and hence there is no problem if the IP address of a unit changes for whatever reason).

To set the link, first set the unit in IP mode (**Setup** / **Network** / **Change Network** / **Ethernet**), and set the desired encoding format (**SETUP** / **Algorithm**, etc.).

Then enter[1] the SIP URI of the unit to call, and press the key.

ⓘ *It is often possible to dial the short form username (omitting the @sipservername) when the device is itself registered onto the same "sipservername" server.*

[1] Use the "#" key to switch the keypad mode numeric/lowercase/uppercase

Receiving calls

This is very simple, in both cases (direct peer to peer link or SIP server). There is nothing to do…

When a call is received, the units negotiate automatically a commonly acceptable coding algorithm, and set the link automatically. On the receiving side, SCOOPY+ will "follow" the calling unit.

ⓘ *Important notice: the unit can receive and accept incoming IP calls even when its current network interface is not Ethernet/IP, provided it not already busy with an established link.*

"Network quality" setting

Depending on the quality of service provided on the network, especially its jitter performance, it is possible to change the stability/latency compromise used by the SCOOPY+. For this purpose, a setting is available in the menu (**Setup** / **Network** / **Ethernet Parameter** / **Network Quality**). Three choices are proposed:

- "HIGH": suitable for a good quality and low jitter network; latency is minimal, but the codec will have little tolerance to possible jitter
- "MIDDLE": intermediate (and default) setting, suitable for a moderate transmission jitter
- "LOW": to be preferred when the network has low QoS, especially for residential ADSL lines. This setting ensures a safer operation, at the cost of a high latency.

On a LAN and/or private network with a controlled quality, the "HIGH" quality setting is recommended, as it yields minimum latency.

On the contrary, it should be avoided for a link via the Internet, as it can only tolerate a low jitter. One solution can be to start with a "MIDDLE" setting, and switch to the "LOW" setting if too much audio disturbance is heard.

Quick redialing

Once a remote device has already been called, it is easy to recall it: press the key, then you can scroll through the call "history" using the arrows. Press again the key when the desired number is highlighted. This is especially efficient for quickly redialling the last called number.

Releasing/Hanging up

To stop a link, press the key.

ⓘ *Press a second time to confirm! (this is for safety, in order to avoid releasing by mistake)*

Links with IP phones

SCOOPY+ is compatible with IP phones that use the SIP protocol (many on the market do). The algorithm used in this case is commonly G711, but a few IP phones also accept G722.

Note that "IP phones" include software SIP phones implemented on computers (also called "softphones") or smartphones.

Multicast mode transmission

On a network that can support it[1], the multicast mode can optimise the resource usage when an audio stream has to be distributed simultaneously to several destinations. Contrary to the normal bidirectional unicast mode, this mode is unidirectional: one *sender* encoder sends a stream towards a multicast *group address*, and one or several decoders receiving the stream pick up the packets sent to this group address and decode the audio stream.

On the SCOOPY+ the operating mode stays quite similar to the "normal" mode, with mainly two differences in the multicast mode:

- A codec must be set as sender or receiver device
- SIP is not used and hence the SIP configuration is not relevant

For the operation it is assumed the network "statically" supports UDP multicast, i.e. routers of the network recognise and deal with routing the packets with multicast group addresses.

For obvious reasons the coding setup is entirely decided at the source. On the codec on the audio source side (hence *sender* of the encoded stream), the procedure for setting up a multicast stream is the following:

- Set the codec in "multicast send" mode. *To do this, refer to the* ***AoIP Parameter*** *sub-menu, page 46.*
- Select on this codec the desired audio coding parameters;
- Start the streaming as for setting up a call in normal mode: enter the IP group address, then press the key.

Note that the codec decodes its own stream on its audio output; this can be useful for audio monitoring.

For each codec that has to decode the stream, the procedure is simple:

- Set the codec in "multicast receive" mode. *To do this, refer to the* ***AoIP Parameter*** *sub-menu, page 46.*
- Start the decoder as for setting up a normal call: enter the IP group address, then press the key.

[1] This does not include the Internet; multicast cannot be used over the Internet.

4.5.4. Setting up a POTS link

Preliminary settings

The network interface must be configured depending on the PSTN line that is used, thanks to the "POTS Parameter" sub-menu (**Setup** / **Network** / **POTS Parameter**)

ⓘ *You should preferably disable services such as call waiting tones. Tone signals injected on the line during the communication have a high risk of disturbing the codec operation, and even dropping the link. It is not recommended to use a line if it is not possible to disable such tones.*

Calling a POTS number

To set the link, first set –if not already done- the unit in POTS mode (menu **Setup** / **Network** / **Change Network** / **POTS**). Then select the desired encoding format (**Setup** / **Algorithm**). Select the desired variation:

- P0: default mode of operation for this
- P1: protected mode 1, with increased resilience to errors (but higher latency)
- P2: protected mode 2, with slightly higher resilience to errors (and slightly higher latency)

ⓘ *Make sure to set units the same on both sides of the link, because this setting cannot be negotiated automatically by the units. It is recommended to use the default P0 mode, the other are mainly for compatibility with older codecs of the AETA product range.*

Come back if needed to the base screen using Esc. Then enter the number of the destination and press the key. The codec dials, negotiates with the remote unit the bit rate (this phase can last several seconds) and sets up the link and the coding.

ⓘ *A message is displayed in case of a failure of the link establishment, or on a remote release. To come back to the base screen, press* Esc *or* .

Receiving a POTS call

Receiving calls is simple. When a call is received, the SCOOPY+ "unhooks" and the link is set up after the negotiation of the bit rate with the calling unit (this can last several seconds).

ⓘ *Important notice: the unit can receive and accept incoming POTS calls even when its current network interface is not the POTS line, provided it is not already busy with an established link. On the other hand, it will ignore such calls if it is running in LL mode, except if the "Backup receive" mode is active.*

Releasing/Hanging up

To stop a link, press the key.

ⓘ *Press a second time to confirm! (this is for safety, in order to avoid releasing by mistake)*

Quick redialing

Once a remote device has already been called, it is easy to recall it: press the key, then you can scroll through the call "history" using the arrows. Press again the key when the desired number is highlighted. This is especially efficient for quickly redialling the last called number.

4.5.5. Setting up a mobile telephone link

Prerequisites

You must have a subscription and a SIM card granting access to mobile transmission. *In the case described here the subscription must include access to the basic telephone service.*

Preliminary settings

First insert the SIM card in the SCOOPY+ (the unit must be powered off for this. See how to proceed on page 34, "Drawer for SIM card *[10]*". Power up the codec, enter the sub-menu **Setup** / **Network** / **Mobile (Int.)** / **Mobile Parameter** / **PIN**. Enter the PIN code of the SIM card using the keypad and press OK. *This is useless, of course, if the card's PIN code is blank or disabled.*
The network access must be configured using the "Mobile Parameter" menu (**Setup** / **Network** / **Mobile Parameter**), see details in page 51, Mobile Parameter.
.

- Select the mobile phone mode: **Mode** / **Cellphone**.
- If needed select the network type: **Network Mode**. In doubt, select "**Auto**" which fits most situations.
- The **Network Operator** sub-menu allows you to choose among the available operators, if your mobile subscription allows you to do so.
- *Other settings (Network Quality, Access Point…) are meaningless for a telephone link.*

Calling a mobile number

To set the link, first set –if not already done- the unit in mobile access mode (menu **Setup** / **Network** / **Change Network** / **Mobile (Int.**). If needed come back to the base screen, using Esc.

Note that the display now shows you the network type (example: 3G) and the radio reception quality. Press Esc *(display the root menu) and you get more details: operator name, more specific technology (example: HSPA).*

Enter the destination number and press .

An error message is displayed in case of a failure of the link establishment, or on a remote release. *To come back to the base screen, press* Esc *or* .

Receiving a mobile phone call

Receiving calls is simple. When a call is received, the SCOOPY+ "unhooks" and the link is set up.

Important notice: the unit can receive and accept incoming mobile phone calls even when its current network interface is not the mobile network, provided it not already busy with an established link. On the other hand, it will ignore such calls if it is running in LL mode, except if the "Backup receive" mode is active (for details see above ***Erreur ! Source du renvoi introuvable.****,* ***Erreur ! Source du renvoi introuvable.****).*

Releasing/Hanging up

To stop a link, press the key.

Press a second time to confirm! (this is for safety, in order to avoid releasing by mistake)

Quick redialing

Once a remote device has already been called, it is easy to recall it: press the key, then you can scroll through the call “history” using the arrows. Press again the key when the desired number is highlighted. This is especially efficient for quickly redialling the last called number.

4.5.6. Setting up a link in IP mode via a mobile network

Prerequisites

You must have a subscription and a SIM card granting access to mobile transmission. *In the case described here the subscription must include access to packet switched data transmission.* It is possible to operate over a 2G network in EDGE mode, if using very low bit rates, but for an acceptable performance it is preferred to access a 3G/3G+ network, at minimum UMTS.

Preliminary settings

First insert the SIM card in the SCOOPY+ (the unit must be powered off for this), and connect at least one antenna to the unit. See how to proceed on page 34, "Drawer for SIM card *[10]*". Power up the codec, enter the sub-menu **Setup** / **Network** / **Mobile (Int.)** / **Mobile Parameter** / **PIN**. Enter the PIN code of the SIM card using the keypad and press OK. *This is useless, of course, if the card's PIN code is blank or disabled.*

ⓘ *If you use a mobile access USB module, insert the SIM card in this device (refer if needed to its dedicated literature) and plug the USB key in the back of the unit. After a while, the module is detected and appears in the list of the network accesses under the name "Mobile 1" or "Mobile 1 (Ext.)". In all the instructions in this chapter, replace "Mobile (Int.)" by this name allocated to the USB device.*

The network access must be configured using the "Mobile Parameter" menu (**Setup** / **Network** / **Mobile Parameter**[1]), see details in page 51, Mobile Parameter.

- Select the packet data mode: **Mode** / **IP Mode**.
- If needed select the network type: **Network Mode**. In doubt, select "**Auto**" which fits most situations. *Avoid giving the priority to GSM, as its performance is poor.*
- The **Network Operator** sub-menu allows you to choose among the available operators, if your mobile subscription allows you to do so.
- Go to the sub-menu **Access Point** / **APN**: enter the APN code of the operator using the keypad, then OK.
- If required enter the other access point settings: user name and password. *This is n ot needed usually, otherwise the needed data are provided along with the subscription and the SIM card.*
- You can define a **Network Quality**: in doubt, select **Middle**. *You can test a setting on a connection with a significant duration, and change the setting if the stability does not look sufficient.*

If a SIP server is used, the SCOOPY+ must be registered on this server. Use the **AoIP Parameter** menu (*see above on page 46*) for entering the SIP account data and activate the SIP registration. *These settings are common to IP modes both via Ethernet and mobile networks.*

Besides, for links over the Internet, most often the link passes through a NAT router which "hides" to the codec the characteristics of the public access: public IP address and RTP ports actually used. This can make the SIP transactions fail, or get a non-optimal route through the network. To avoid this a STUN server is often used; the corresponding settings are also found in the **AoIP Parameter** sub-menu. *These settings are common to IP modes both via Ethernet and mobile networks.*

ⓘ *You can find in annex 7.3 (Some methods to deal with NAT routers and firewalls) some additional information on the use of STUN.*

[1] Or "Mobile 1" for an external USB device

Calling via a mobile IP network

To set the link, first set –if not already done- the unit in mobile access mode (menu **Setup** / **Network** / **Change Network** / **Mobile (xxx)**[1]. If needed come back to the base screen, using Esc.

ⓘ *Note that the display now shows you the network type (example: 3G) and the radio reception quality. Press* Esc *(display the root menu) and you get more details: operator name, more specific technology (example: HSPA).*

Enter the destination number and press .

From this moment you can set a call just like a call over the Ethernet interface (see above in *4.5.3, Setting up a link in IP mode via* Ethernet):

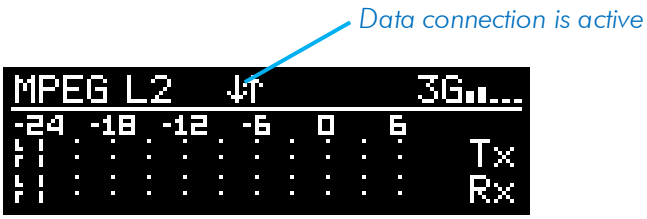
- Call directly, without a SIP server: enter the remote device IP address and press the key.
- Use a SIP server: enter the SIP URI of the remote device and press the key.

An error message is displayed in case of a failure of the link establishment, or on a remote release. *To come back to the base screen, press* Esc *or* .

Receiving a mobile IP call

Receiving a call is basically the same as through the Ethernet interface, but with some significant differences:

- SCOOPY+ must be connected to the mobile data service to be able to receive requests from the network (such connection is somewhat equivalent to the physical connection of an Ethernet cable). This connection is performed automatically by SCOOPY+ just before setting an outgoing call. To make it "manually" and get ready to receive a call, you must use the sub-menu **Setup** / **Network** / **Mobile Parameter / Data Connection**. *The "Connected" status is recalled on the base screen and the main menu*:



- Mobile networks often include devices blocking the access *towards* a mobile terminal: NAT routers, often symmetrical (which prevents a STUN server to be efficient), etc. Moreover, you rarely get a public IP address, and it is never possible to get a public and *fixed* IP address.

For these reasons it is rather difficult to receive a call without using a SIP server. Conversely, if the SCOOPY+ is registered on a SIP server, once the data connection is active the codec is actually able to receive a call.

When a call is received, the units negotiate automatically a commonly acceptable coding algorithm, and set the link automatically. On the receiving side, SCOOPY+ will "follow" the calling unit.

ⓘ *Important notice: the unit can receive and accept incoming IP calls even when its current network interface is not mobile IP, provided it is not already busy with an established link.*

Releasing/Hanging up

To stop a link, press the key.

[1] The actual name of the network access depends on the concerned module: "Mobile (Int.)" for the module integrated with the "wireless" option, "Mobile 1 (Ext.)" for an external USB device, etc.

Press a second time to confirm! (this is for safety, in order to avoid releasing by mistake)

Quick redialing

Once a remote device has already been called, it is easy to recall it: press the key, then you can scroll through the call "history" using the arrows. Press again the key when the desired number is highlighted. This is especially efficient for quickly redialling the last called number.

4.6. Management of the configuration profiles

The configuration profiles ease configuration changes and the setting up of routine links. There are three types of profiles:

- "**Remote profiles**", which include call parameters for a given **destination** (number(s), coding, etc.). Such a profile is similar to a directory entry, but in addition it can memorize the coding parameters.
- "**Local profiles**" memorize the **network access** parameters. These are the settings found in the sub-menu **Setup** / **Network**. Recalling a local profile is a quick way to restore the configuration needed for a given line or network.
- "**Snapshots**" memorize the settings for the **audio** interfaces. These are settings found in the **Setup** / **Audio** sub-menu.

These various memories are usable locally, but also through the embedded html pages, and also using these pages they can be imported/exported from/to a computer.

Each remote profile includes the following elements:

- *Name* assigned to the profile
- *Number(s)*: ISDN, POTS or mobile call number(s), or IP URI, Ip address, depending on the transmission mode.
- *Coding configuration* associated with the profile

A remote profile may have no number included; in such case it is used for quick and safe recall of a given coding configuration.
Conversely a remote profile may have no associated coding configuration included; in such case it is just like a directory entry, used for instant dialling a known destination.

The profiles are managed and used via the "Profiles" menu (refer to details in 4.4.4, **Profiles** menu). For remote profiles, the available possibilities are:

- Create a *New* profile, assigning it a name, possibly call number(s), and possibly an associated coding configuration.
- *Load* a profile previously recorded: the codec then configures itself with the coding configuration memorized in the profile.
- *Call* directly a profile: the codec then configures itself with the coding configuration memorized in the profile (if any), and sets a call to the number(s) included in the profile.
- *Delete* a profile previously recorded.
- *Update* profile previously recorded, either for renaming it or change its content.

For local profiles or snapshots, the available possibilities are:

- Create a *New* profile, which will memorize all the current network settings (for a local profile) or all the audio settings (for a snapshot), assigning it a name.
- *Load* a local profile or snapshot: the codec then restores the settings included in the profile.
- *Delete* a local profile or snapshot.

4.7. Restricted operation mode

4.7.1. Principles

SCOOPY+ can be locked in a restricted user access mode, which limits user access to the essential controls and features. For routine operation, this mode allows to avoid risks such as random settings, with unpredictable effects, handling or adjustment mistakes, accidental erasure of useful memories, etc.

When SCOOPY+ is in restricted access, it is still quite operative, especially if suitable memories (profiles, snapshots) have been prepared beforehand, because the following functions are still accessible:

- Receiving calls
- Releasing links
- Redialing (recalling previous numbers) outgoing calls
- Loading and calling remote profiles
- Loading local profiles or snapshots

On the other hand, it is not possible to change settings via the menus, as only the "Profiles" menu is accessible. It is also impossible to delete or edit the recorded profiles.

A password is needed for switching to the restricted mode (locking) or back to full access (unlocking). This password can be programmed (of course, not when in restricted access...).

Warning: this access protection is completely independent from the one that can be set for the "web" interface (embedded html pages). The passwords have no relationship, and are set separately. One of the interfaces (front panel or html) can be locked while the other is not.

4.7.2. Locking the front panel

Managing the restricted mode is done with the sub-menu **Tools** / **Maintenance**. See in 4.4.7 (Maintenance sub-menu) this sub-menu items.

First you must program the desired password, because the password is blank when the unit comes from the factory or after a complete "factory reset". The password is an arbitrary sequence of numbers.

For switching to the restricted mode, select **User Access** and confirm (**Yes**). You must then enter the password to actually lock the unit (this makes sure you really know the password before locking). The screen warns you about the switching to restricted access.

Press Esc and come back to the base screen. From then on, on every attempt to navigate in the non authorized menus, the following message is displayed:

4.7.3. Unlocking the front panel

From the base screen or the root menu, type the password sequence: SCOOPY+ unlocks:

The full access is restored. Press Esc to come back to the base screen.

4.7.4. Lost the password?

You have forgotten or lost the password ? The only way to restore full access is to clear all the settings, which will also clear the passwords. The unit will be accessible again.

Chapter 4.8 below shows the procedure for such erasure.

ⓘ *In order not to lose all the memorized settings, you can perform a complete reset of the settings without deleting the profiles. These can be useful for you to come back more easily to the desired configuration.*

4.8. Clearing all settings

In some cases (especially if the password is lost), it can be necessary to come back to the "factory" configuration.

To clear up all the configuration and come back to the factory settings, normally you should go to the sub-menu **Tools** / **Misc** / **General Reset** and select **Factory Reset**, or **Reset Settings** (which does not delete the memorized profiles and snapshots).

The default password is blank. *In the factory setting, menus use the English language. Other languages can be selected via **Tools** / **Misc** / **Language.***

ⓘ *If it is not possible to access the menu to perform the factory reset (password lost), the erasure should be done using the embedded html pages: see chapter 4.11 below for the operating mode.*

4.9. Backing up and restoring the configuration

It is possible to save all the equipment settings in a file, and conversely to restore a complete configuration from a file previously recorded in such way. Such transfers can be done using the embedded html server; refer to 5.10.6 for the operating mode.

Partial exports are also possible, selecting elements to include in the export: remote profiles, local profiles, snapshots, settings…

4.10. Displaying received SMS

When the unit is equipped with the "wireless" option and it is registered on a mobile network, SCOOPY+ can display SMS received on the SIM card's subscriber number. The operation is very simple: on receiving the SMS the text is directly displayed on the screen:

Whenever the complete message cannot be displayed on the screen, use the arrows to scroll the message and read all the text. After the message is read, press Esc to come back to the normal display.

Caution: the message is now cleared and it cannot be displayed again.

ⓘ *If you do not want to display the received SMS, you can disable this feature: go to the sub-menu **Tools** / **Misc** / **Aux. Functions** and disable the **SMS** parameter.*

4.11. Audio File (record)

This is an option on the Scoopy+.

Scoopy offers a module to record what goes into the unit to read a file already recorded in the device and edit a file to cut it. The module also allows you to send files via FTP and export files in the cut recording format.

The recorder save audio file on the SDCARD. The format of the file is mono or stereo audio wav file (See Audio menu to set it). Records are managed by the top keypad.

1. Start Marker
2. 10s Backward
3. Play / Pause
4. Stop
5. Record
6. 10s Forward
7. End Marker

4.11.1. Settings

In the Setup menu, 'Audio', 'Recording Mode', it is possible to choose to save either in mono or stereo the mix of inputs.

WARNING: When you enable the coordination channel, the recorder records the main program and the coordination return channel. This configuration is perfect to record a remote interview. See Routing chapter for more details.

4.11.2. Record

A first press on REC button starts buffering for 30 seconds. A second REC press start recording. To pause the recording, simply press PLAY. By pressing it again, the recording resumes. Recording stops by pressing the STOP button.

The file is placed in the Recordings folder. It is directly accessible from the PLAY button or the Directory menu from the main menu. It is named default 'record_xxx' and it can be renamed via the menu available by selecting the file 'OK' and then selecting the sub menu renamed.

4.11.3. Play

To read the file, go to the Records menu and then select the file to play and then press either PLAY or through the sub-menu, click 'Play'. You can navigate through the file with Backward and Forward keys. Respectively allow it to back and forward 10 seconds. To pause press PLAY, to stop playback press STOP.

CAUTION: While playing a file, it is not possible to adjust the output level.

It is possible to know the duration of the recording, Track mode and file size. From the menu record, after selecting the desired file, click the Browser button to the right to show all details.

4.11.4. Edit

To split a file, go to the Directories menu, select your file, press play. Press F1 to mark the beginning of the cut and then press F2 to mark the end of the cut. You can cancel brands by re-pressing the F1 or F2 button.

By pressing the STOP button, a screen will appear asking you if you want to save the location of the marks, selecting 'yes', the screen will ask you to type the new name for the file. You can find this file in 'Taken' via directories or by pressing the play button. You can always change brands by opening the file ('Play' or

press the play button).

To read the file from the Records menu and have a wave file that can be sent via FTP format, it is recommended to export the audio file format. To do this go to the 'Takes' menu, select the file then press 'OK' select 'Export'.

Enter the name of the file and save. Your file is now visible in the Records menu.

5. Operating mode – Embedded HTML pages

The embedded html server in SCOOPY+ provides a comfortable and efficient means to control and monitor the unit. It just needs the SCOOPY+ to be connected to an IP network and to be reachable from a computer, or another device with an html browser: tablet, smartphone…

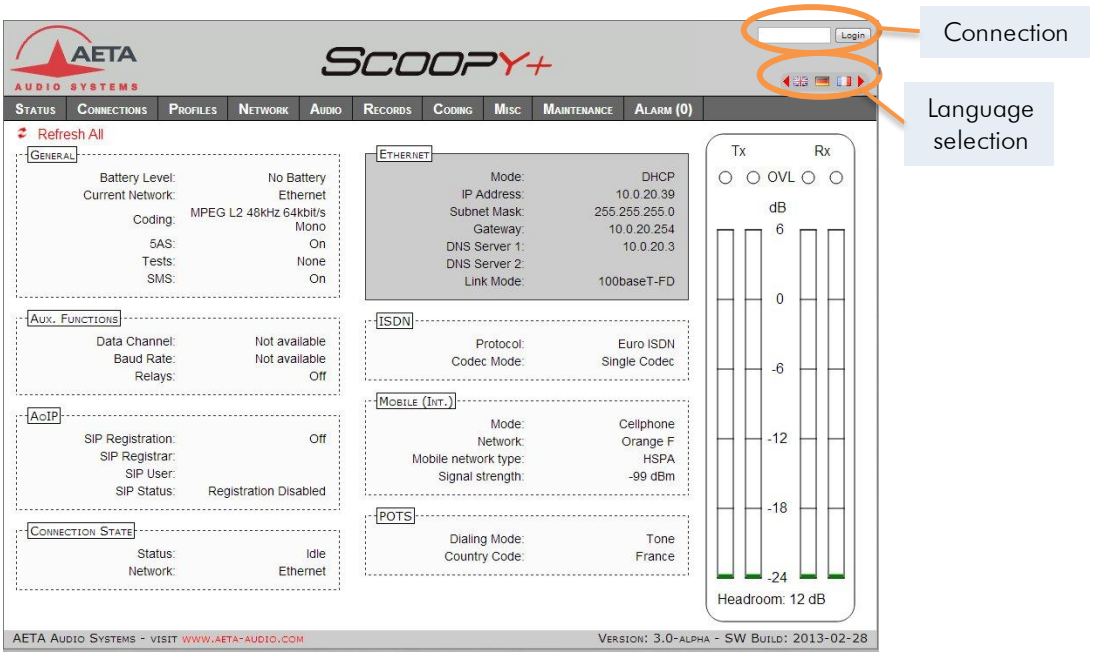
In the most common case, the two devices are connected on the same local network (LAN). But it is also possible to control the SCOOPY+ by remote, provided that the control device can reach it (TCP/IP port 80, HTTP protocol).

If you also have a secondary Ethernet interface (using a USB/Ethernet adapter), each of the two interfaces can be used for this remote control access.

This control mode is usable regardless of the OS of the control unit, and the embedded pages are compatible with all common browsers. No software installation is needed on the control position.

5.1. Accessing the SCOOPY+ html pages

Once the SCOOPY+ is connected on an IP network, the first step is to get its IP address, from the menu: **Tools** / **About** / **Local IP**[1]. Then, on the control device, launch the html browser and enter the IP address of the SCOOPY+ in the "address" or "URL" field. This gives access to the html server integrated in SCOOPY+. The page which is displayed is similar to the following picture:



If needed, select another language by clicking the suitable flag (*this choice is not linked to the language selected for the menus on the front panel interface*).

The home page displayed above is the "**Status**" page, which provides an overview of the unit status, but allows no action on it. This is the only "free access" page, with no limitation or access control.

To access the other pages, you must "log in", and get for the control device an exclusive access. Any connection request from another device will remove this access.

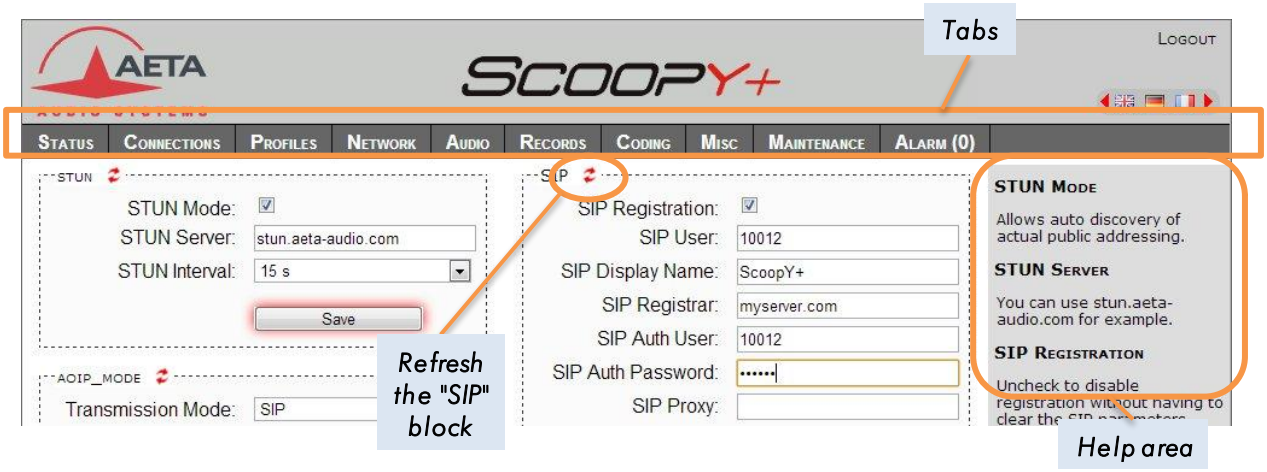
To log in, enter the password and click the connection button. The initial password is blank: click directly on the button. To set the password and enable protection, go to the "Maintenance" page (cf. further).

To release control, click on "Logout" (also in the connection area). You are also logged out automatically after a long period of time with no action on the pages.

[1] For a possible secondary Ethernet interface, menu Config / Network / Ethernet 1 Parameter...

5.2. Principles of operation with html pages

The picture below shows a typical page.



On top you find a bar of tabs corresponding to the various categories of functions and parameters for the SCOOPY+. Clicking a tab you access either a page, or a drop-down list for selecting a secondary page. *These tabs and pages are detailed in the following chapters.*

Under the bar can be found the information and adjustable parameters, with various selection or entry modes for these parameters, grouped in blocks (each surrounded with a frame). On the right side, a text area provides additional help and hints.

As a general rule, the displayed parameters are read when accessing the page, and are not refreshed automatically[1]. To force a refresh, click the icon: the data in the area or frame are read and refreshed.

ⓘ *Exception: some data on some pages is however periodically and automatically refreshed. This makes a modest bit rate, but you should exit the html pages if you want no traffic at all on the path between SCOOPY+ and the control device.*

One tabs have a specific behavior:

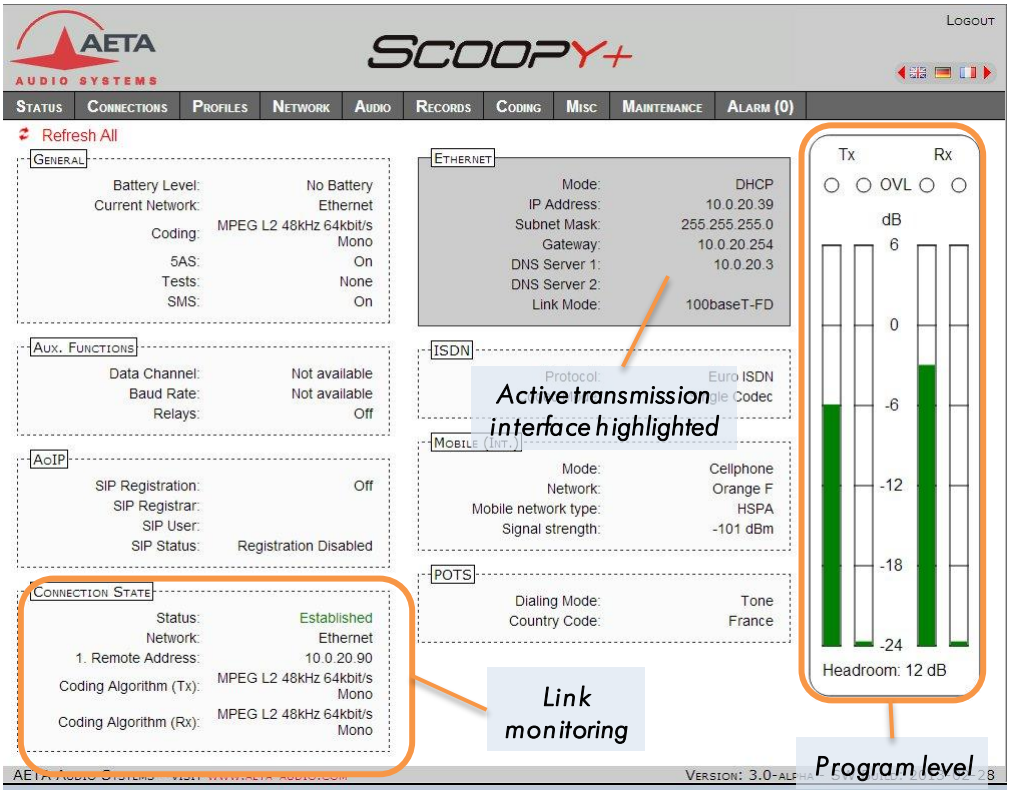
- "Status" is accessible without a login and some data are updated automatically.

Access to the tabs (other than "Status") requires loggin in beforehand. If you click a tab without being logged in, the login dialog box opens to allow you to enter the connection password. If it is blank, just click the "Login" button.

[1] This is on purpose, in order to avoid a permanent high rate of queries to the unit, which may be questionable in some situations.

5.3. "Status" tab

This tab leads to the "**Status**" page, which is also the home page. It provides an overview of the essential settings and parameters, sufficient for basic monitoring.



This page is accessible without a log in, and without a password.

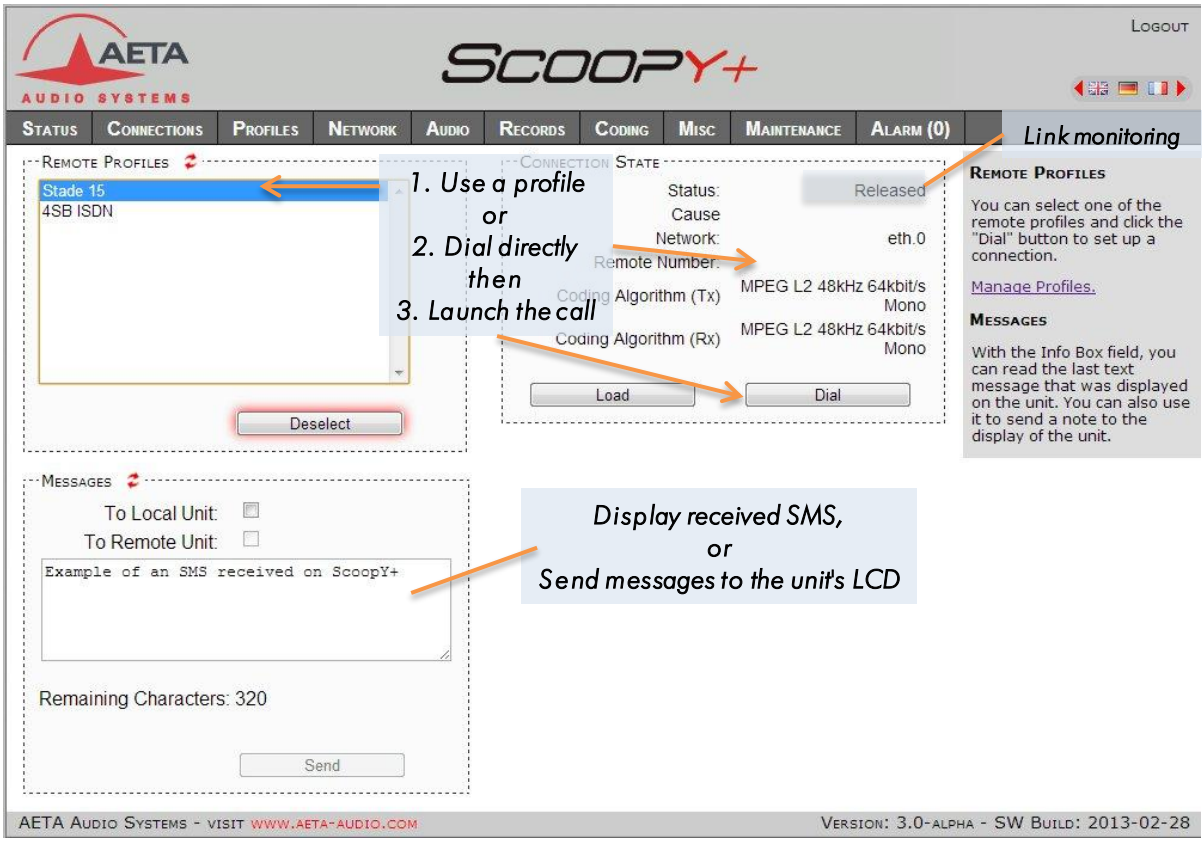
The link status is monitored dynamically; e.g. you can see call reception and releasing. It is also possible to monitor calls that an operator is managing using the front panel.

The page displays the audio level of the two transmitted signals (encoder) and the two received signals (decoder), shown as bargraphs with a 0 dB reference. *Refer to chapter 4.4.3 (Audio sub-menu) for more details on the setting for this reference. The "Headroom" setting is adjustable using the "Audio" tab.*

ⓘ *Note: the goal of these bargraphs is to provide an indication on the presence and level of the audio modulation. The measurement has a good precision for stationary signals (1 dB accuracy and resolution), but the update rate is rather slow. As a consequence these bargraphs are not appropriate for precisely monitoring a dynamic program, and they should not be considered an alternative to real program meters!*

5.4. "Connections" tab

This tab leads to the "**Connections**" tab, allowing to remote control the links: monitoring, outgoing call set up and call releasing.



The "Connection State" block allows to monitor and/or manage links.

- For direct dialing: the codec has to be set beforehand for the right interface (see "**Network**" tab), and the suitable coding configuration ("**Coding**" tab). Enter the number (or numbers, or SIP URI…), then click "Dial". *You can also use one of the last called numbers: click the arrow on the right of the "Number" field and pick a number in the dropdown list.*
- For using a remote profile: select the profile in the list on the left, and click the "Dial" button. Alternatively, you can click "Load". The codec loads the profile settings but does not dial: this is useful for quickly setting the coding parameters.

Indicators show the status of the relays when the "Relay transmission" feature is active. *For auxiliary functions, see the "**Coding**" tab.*

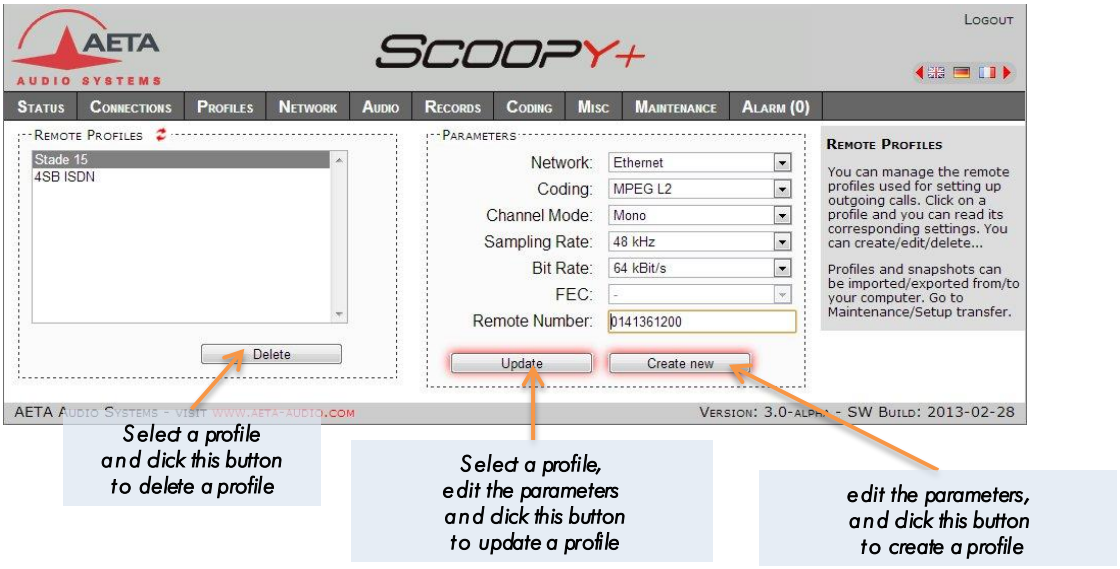
The SMS received by SCOOPY+ are displayed in the "Messages" frame (in addition to being displayed on the unit's LCD). This block can be used also to send a text to the LCD from the control position.

ⓘ *When SCOOPY+ is in double ISDN codec mode, the "Connection State" frame is replicated, one frame for each codec. But selecting a remote profile switches back to the normal display (for the single codec), because the profile data correspond to a single codec (remote profiles cannot be used in double codec mode).*

5.5. "Profiles" tab

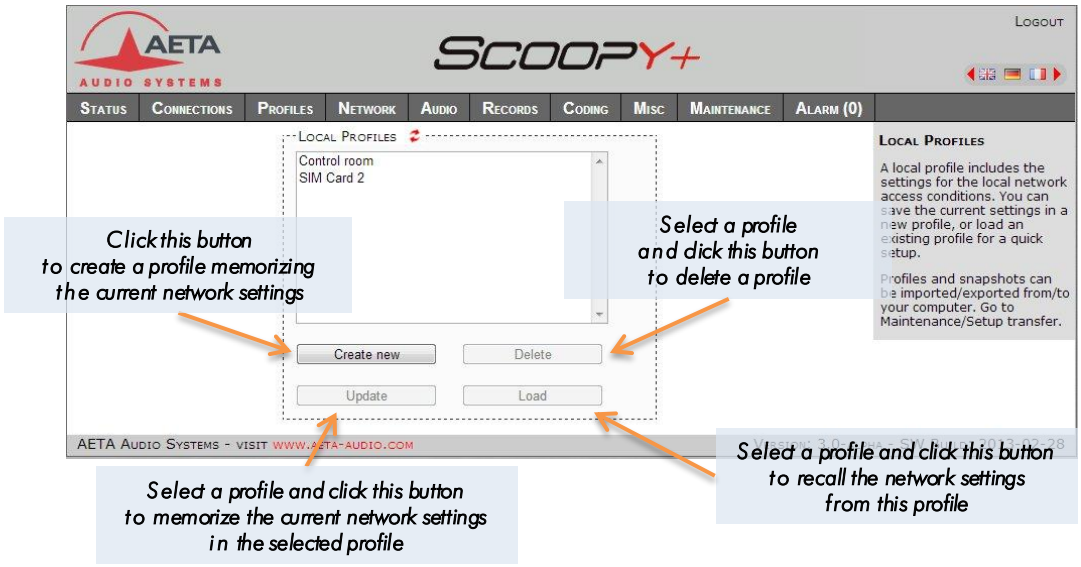
This tab proposes three secondary pages, one for each profile type.

5.5.1. "Remote profiles" page



Parameters that are not relevant in a given context are shaded and inactive.

5.5.2. "Local profiles" page


AETA AUDIO SYSTEMS
SCOOPY+
LOGOUT
STATUS
CONNECTIONS
PROFILES
NETWORK
AUDIO
RECORDS
CODING
MISC
MAINTENANCE
ALARM (0)
LOCAL PROFILES
Control room
SIM Card 2
Create new
Delete
Update
Load
LOCAL PROFILES
A local profile includes the settings for the local network access conditions. You can save the current settings in a new profile, or load an existing profile for a quick setup.
Profiles and snapshots can be imported/exported from/to your computer. Go to Maintenance/Setup transfer.
AETA AUDIO SYSTEMS - VISIT WWW.AETA-AUDIO.COM
Click this button to create a profile memorizing the current network settings
Select a profile and click this button to delete a profile
Select a profile and click this button to memorize the current network settings in the selected profile
Select a profile and click this button to recall the network settings from this profile

5.5.3. Snapshots

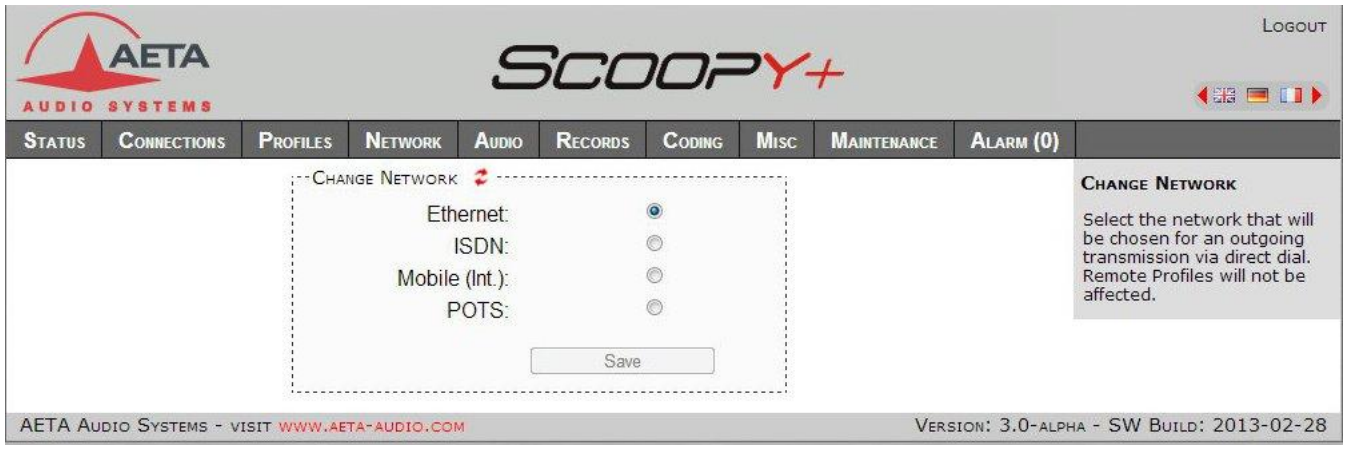
Snapshots are in fact managed on the "Audio" page, and this "Snapshots" choice on the "Profiles" tab just redirects to the "Audio" page. *See further the description of the audio page.*

5.6. "Network" tab

This tab leads to several pages, which correspond, just like the sub-menu "**Network**" of the front panel interface (see 4.4.1 above), to the various network interfaces. For this reason, the number of pages actually proposed depends on the version and the options of the unit (with or without ISDN, mobile access, etc.).

5.6.1. "Change Network" page: default interface selection

This page is used for selecting the default interface: this is the one that is implicitly used when dialing to set an outgoing call. Besides, codec algorithm settings (either via menus or the html pages) are applicable to this interface[1].



Select the desired interface and click "Save".

5.6.2. "Ethernet Parameter" page

More details on these parameters in 4.4.1, Ethernet Parameters.

Remember to **save the changes**, clicking the "Save" button!

[1] Remind that the algorithm settings are memorized separately for each network interface. When switching the default interface, the codec settings for this interface are recalled.

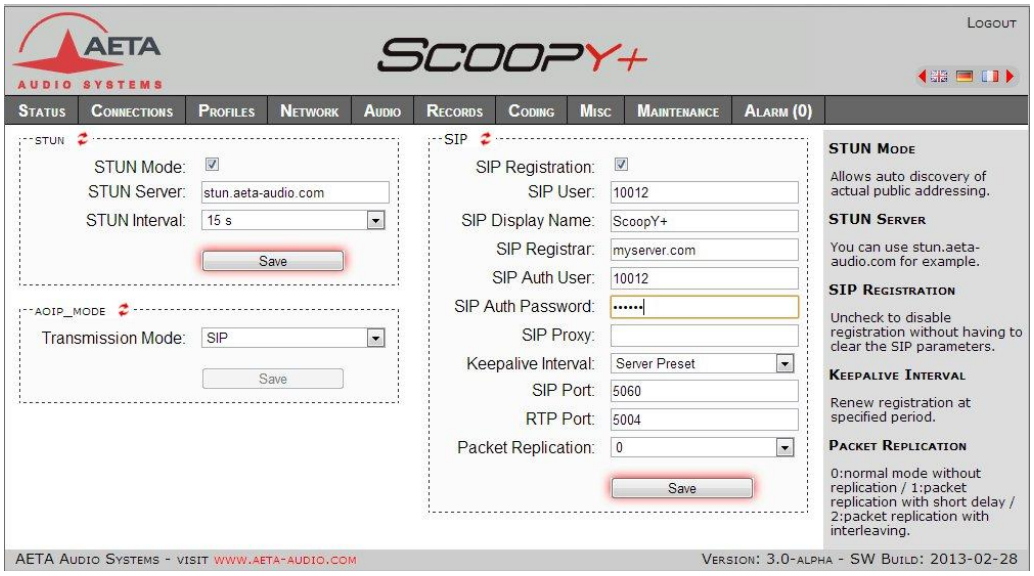
ⓘ *Settings regarding the Ethernet interface and IP addressing are especially critical, because this is just the interface you are controlling SCOOPY+ through. You can lose control because of a mistake in the settings; it might even be the normal expected consequence for a configuration change. So be very careful with the settings on this page!*

In case control is lost after an error, you must regain control and correct the settings using the front panel of SCOOPY+.

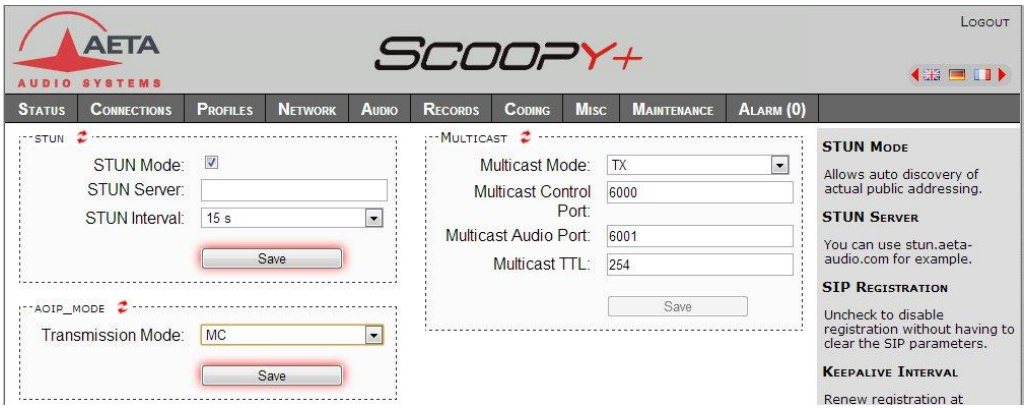
Note: if a secondary Ethernet interface is available, it is also shown in this page, in an "Ethernet 1" frame.

5.6.3. "AoIP Parameter" page

More details on these parameters in 4.4.1, AoIP Parameter.



If you select (in the "AoIP Mode" frame) the multicast mode, the page is different:



Remember to **save the changes**, clicking the "Save" button!

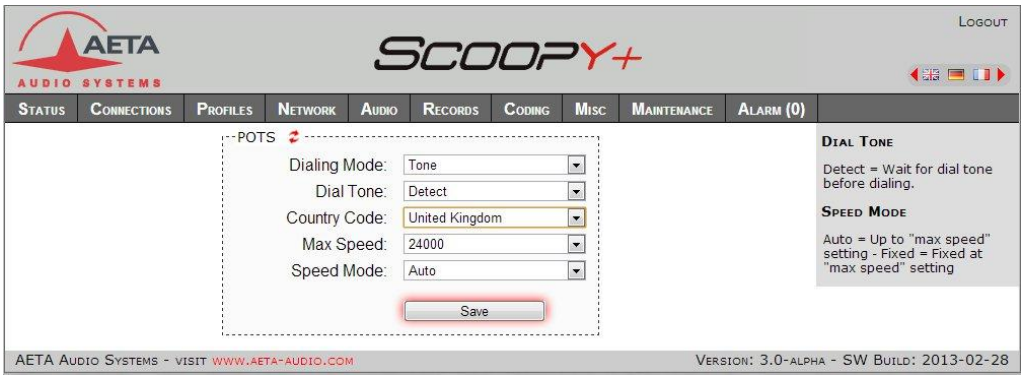
5.6.4. ISDN Parameter page

This page is only visible for units equipped with ISDN interface(s). *See more details on the parameters in 4.4.1, ISDN Parameter.*

Remember to **save the changes**, clicking the "Save" button!

5.6.5. "POTS Parameters" page

This page is only visible for units equipped with the POTS option. *See more details on the parameters in 4.4.1, POTS Parameter.*



Remember to **save the changes**, clicking the "Save" button!

5.6.6. "Mobile Parameter" page

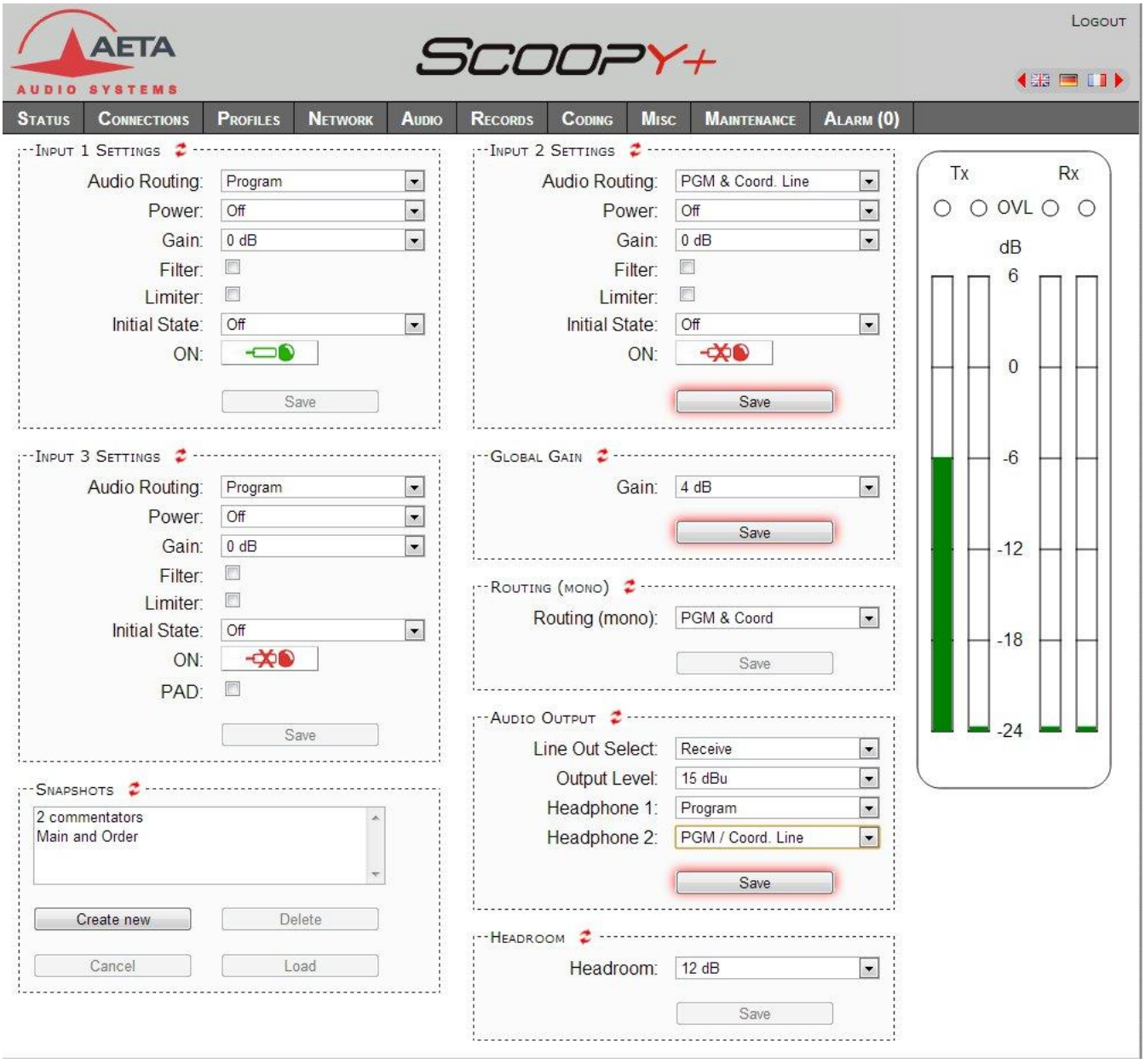
This page is only visible for units equipped with the "wireless" option. See more details on the parameters in 4.4.1, Mobile Parameter. You cannot access all settings unless the PIN code has been entered (otherwise the view is partial only).

ⓘ *If you have plugged a USB mobile access device, it is also assigned a specific parameter page.*

Remember to **save the changes**, clicking the "Save" button!

5.7. "Audio" tab

See all details for these parameters in 4.4.3, Audio sub-menu. Like the "**Status**" page, this page shows the audio level transmitted/received to/from the network.



After changes on the audio parameters, make sure to **save the changes**, clicking the "Save" button.

This page includes the management of *snapshots*, memorizing the parameters controlled in this page.

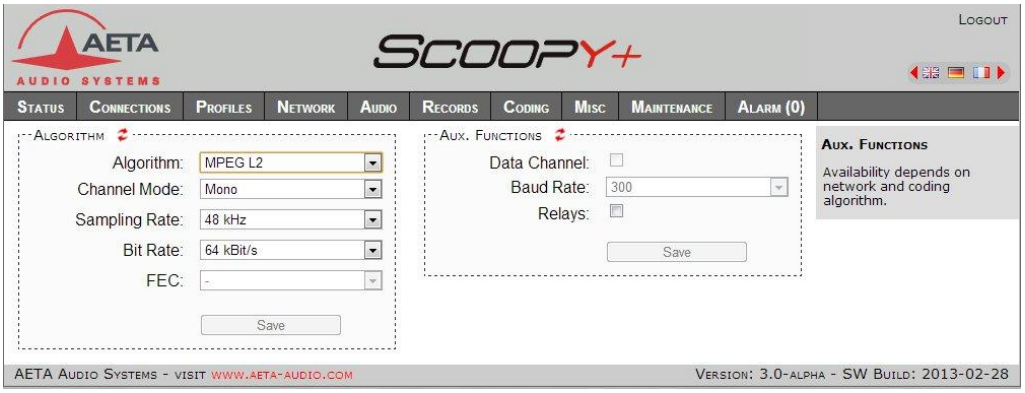
To **create** a snapshot based on the current settings, click the "**Create new**" button, and enter a name for the snapshot.

The other buttons can be used after selecting an existing snapshot. Click a snapshot: its parameters are recalled in the fields of the page, but yet no change is done (the "Save" buttons warn of this). Then you can:

- Click the "**Cancel**" button to simply come back to the previous situation (*also possible using the icons, but one frame at a time*).
- Click the "**Load**" button and directly apply the snapshot (*also possible using the "Save" buttons, but one frame at a time).*
- Click "**Delete**" to remove the selected snapshot from the list of snapshots.

5.8. "Coding" tab

The "Coding" tab groups the selection of the algorithm and its parameters, as well as the auxiliary functions. See details on all these parameters in 4.4.2, Algorithm sub-menu and 4.4.6, Misc sub-menu.

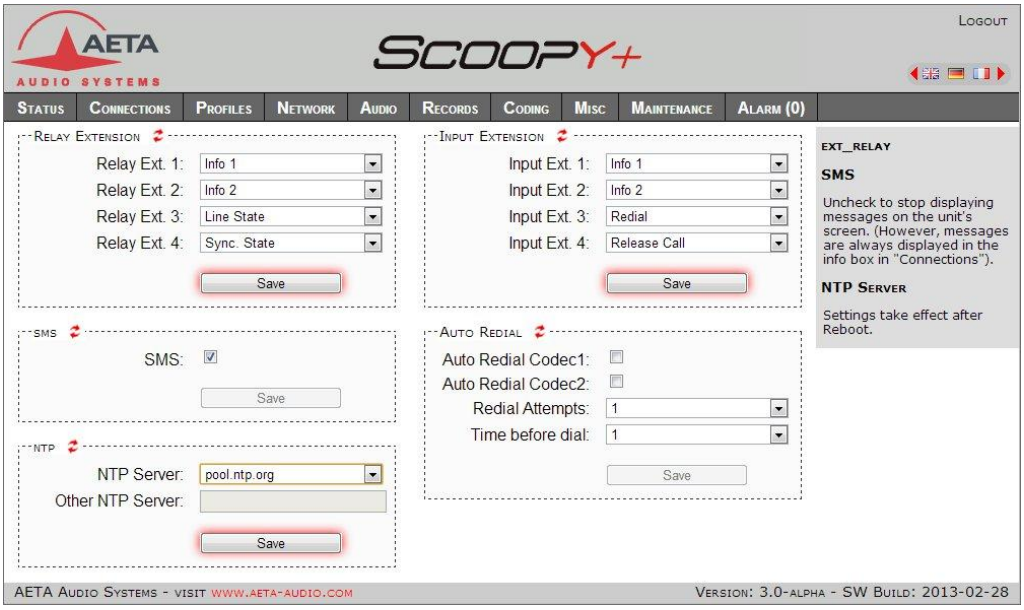


The available coding algorithms depend on the current network interface and the installed options, such as e.g. the AAC codecs. The coding parameters depend on the algorithm and the network interface. This is also true for the auxiliary functions.

Remember to **save the changes**, clicking the "Save" button!

5.9. "Misc" tab

This Misc tab groups several settings that can be found in the Misc sub-menu of the front panel user interface (see 4.4.6, Misc sub-menu): auto redialing configuration and activation, loop control activation, SMS reception enabling/disabling…



It is also possible to configure a NTP server: this kind of server provides a time reference (from the Internet or a server on the local network), useful and recommended for the absolute time stamping of the events in the log (see further 5.10.5, "Event log" page). If such a server is available and accessible via the Ethernet interface, enter its address in the "NTP Server" field. A few public servers are also proposed in the dropdown list[1].

ⓘ *A reboot is needed for this setting to become effective!*

ⓘ *Remember to* **save the changes**, *clicking the "Save" button!*

[1] Note: to be able to use these servers 1) access to the Internet from the codec must not be restricted, 2) a DNS must be set so the symbolic names can be resolved.

5.10. "Maintenance" tab

This tab leads to a selection between several pages dedicated to maintenance aspects.

5.10.1. "Login data" page

This page is for setting the password for logging in the html pages. In a classic way, to set up a password you must first enter the current one, and then enter the new one and confirm it.

Remember to **save the change**, clicking the "Save" button!

ⓘ *Reminder: this password has no relationship with that for unlocking the restricted access on the keypad/display front panel interface.*

If you lose the password: using the front panel interface of SCOOPY+, perform a complete reset of the settings, or a complete "factory reset". The passwords (front panel and html access) are then reset as well as the other parameters in the unit.

5.10.2. "Tests" page

This page provides the same choice as the **Maintenance** / **Tests** sub-menu (see in 4.4.7 and 2.5.5, Test functions, the description, of the various test loops available).

ⓘ *Do not forget to disable the test loops (setting "None") to come back to normal operation!*

5.10.3. "System update" page

This page allows to upload a system update file into the unit and update the unit's firmware.

For an update, the file must be provided to you by AETA (or possibly your distributor/dealer). The procedure is rather simple:

- From this page, click "Browse…", find and select the update file.
- Click the "Update" button.
- The process begins, and a warning message is also displayed on the SCOOPY+ front panel.
- Wait for the completion of the update. Normally the unit should reboot by itself at the end of the process.
- Once the unit has restarted, perform a complete "Factory reset".

This procedure requires maximum care, because such update always implies some risk. Please note these additional recommendations:

- Beforehand backup/export the recorded profiles and snapshots, if you would like to use them again afterwards.
- Make sure the link between the control computer and the SCOOPY+ is stable (no undesirable interruption during the process).
- Make sure the SCOOPY+ is not powered off and its Ethernet interface is not disconnected during the process, do not use its keypad during the process.
- Re-import your profiles and snapshots after the update. *But be aware that, although we always try to keep a maximum upward compatibility, AETA cannot guarantee the re-usability and consistency of these memories after a firmware update.*

5.10.4. "Reset" page

This page provides two functions: clearing the settings and/or memories/profiles recorded in the unit, and rebooting the unit.

You can reset the unit completely or only clear a category of data; the available options are:

- "**Factory reset**": all parameters are brought back to their default setting, and all memories (profiles, snapshots) are deleted.
- "**Reset settings**": all parameters are brought back to their default setting, but all memories are kept: profiles and snapshots.
- "**Reset snapshots**": all snapshots are cleared form the memory.
- "**Reset profiles**": all profiles (remote and local) are cleared form the memory.

The page also allows you to reboot the unit by remote, clicking the "**Reboot**" button. The device is actually reboot when you confirm by clicking "OK" in the dialog box which then shows up[1]. This is equivalent for SCOOPY+ to an off/on power cycle. Of course, you must wait for the return to the normal operative state before regaining control via the html pages.

[1] If you mistakenly clicked the "Reboot" button, it is still possible to cancel this: close the page without clicking "OK" (do not hit the computer's *Esc* key neither).

5.10.5. "Event log" page

This page displays a history if the system events, which can be useful for operational records, or for investigating operation issues, etc.

Events are logged by the codec in its internal memory card; they are recorded in plain text (unformatted ASCII), and the web page displays the 200 last lines of this log (with a scroll bar).

You can select the detail level for this history, depending on the target application:

- "Coarse": only essential events are logged, such as reboots, link set up, alarms… This type of history can be useful as a link record, and gives a very synthetic overview.
- "Normal": gives more details, and this is the default setting, suitable for most operation needs.
- "Debug": absolutely all events are logged; this makes a very "verbose" and technical history, reserved for testing and fixing issues.

SCOOPY+ can also send event messages to a SYSLOG server is such server is available on the network: enter the address or name of this server and click the "Apply" button. From then on, SCOOPY+ sends the designated server all events, regardless of the detail level selected as described above. The two "histories" run in parallel: on one hand messages to the SYSLOG server, on the other hand the "filtered" events (depending on the detail level selected) recorded in the SCOOPY+ log file.

Events are time-stamped (date, hour, minute, second) from the internal clock of the unit. This clock is not backed (no battery inside the unit), but the unit can synchronize at boot time using the NTP protocol. The address of such server must be set in the "Misc" page (see 5.9 above).

Important notice: time stamps are universal time (UTC), hence they take no account of geographical location, neither any DST (daylight saving time). Remember to add the appropriate time difference when analysing the logs.

Under the log window you can find some control buttons:

- "Hold": normally the event window scrolls at any time while new events occur. Click this button to stop this and freeze the display. The button becomes "Continue". *The events are still logged, only the page update is interrupted.*
- "Continue": resumes the automatic updating of the event display.
- "Refresh": click this button to update the display and show the most recent events. This button is relevant if the display has been frozen (with "Hold"), in order to refresh the event display on demand.
- "Save logfile as": allows you to download on the control computer the history file. This takes place as a typical download. Depending on the browser and its settings, you can possibly select the file destination, name…

5.10.6. "Setup Transfer" page

This page provides a means to backup all the settings of the unit into a file, and reciprocally to restore a complete configuration from a file previously saved in this way.

Partial exports are also possible, selecting the items you wish to include: remote profiles, local profiles, snapshots, settings…

The left part of the page is dedicated to "exporting" the device configuration: the configuration is then saved in a file, downloaded in the control computer. Before doing the export you can select which type of data is to be included in this backup/export: check the desired categories (multiple choice allowed), and click "**Export**". The rest is carried out like a typical download. Depending on the browser and its settings, you can possibly select the file destination, name…

The right side is for "importing" the device configuration: click "**Browse…**" to select the file to be imported, and launch the process by clicking "**Update**". The file must be a file previously exported from a SCOOPY+ (same unit, or another). The settings or memories included in the file are respectively applied to the codec, or recorded in its memory of profiles/snapshots.

ⓘ *Warning: the export files can be imported into a device with the same firmware version, but there is no guarantee of "portability" of the configuration files from one firmware version to another. In other terms, the outcome is not guaranteed when importing into a unit files which were exported from a unit with a different firmware version. We recommend to carefully check the settings in such case, after importing the file.*

5.11. "Alarm" tab

The "Alarm" tab switches to red as soon as at least one anomaly is detected. In addition the number of abnormal conditions is shown, like in the example below with two alarm conditions detected:

Clicking the tab, you reach this "Alarm" page showing all the alarm conditions. All possible issues are listed, but only those actually detected are highlighted with the icon.

The issues are grouped in three areas:

- Alarms related to "Transmission": essentially issues related to the decoder synchronization. In addition, for ISDN links, there may be encoder or decoder fallback cases ("fallback" is the situation when the encoder or decoder configuration actually set on the link differs from the one which was initially programmed or expected).
- Alarms related to the audio interfaces: audio clipping (minor alarm), AES input errors.

6. Technical characteristics

6.1. Characteristics of interfaces

6.1.1. Analogue audio inputs

Audio characteristics are measured over a 20 to 20 000 Hz bandwidth except when differently stated.
The inputs are balanced type, using 3-pin female XLR sockets.

Microphone powering:	Phantom 48V or 12V, optional Tonadder 12V
Maximum input level:	+19 dBu (+39dbu Channel 3)
Input stage sensitivity adjustment:	+0 to +48 dB by steps of 16dB
CMRR :	>80dB @ 1kHz
Nominal input impedance:	10 kΩ

6.1.2. Analogue audio outputs

Audio characteristics are measured over a 20 to 20 000 Hz bandwidth except when differently stated.
The outputs are balanced type, using 3-pin male XLR sockets.

Maximum output level:	adjustable from -11 to +22 dBu ± 0.3 dB
Nominal load impedance:	600 Ω or 10 kΩ
Output impedance:	<50 Ω
Symmetry:	> 40 dB (ZL = 150 Ω)

6.1.3. Headphone outputs

Theses outputs (6.35 mm jack on front panel) are for the connection of a 32 Ω headphone. It is also possible to plug a high impedance headphone; however, the maximum available power will be lower.

6.1.4. Ethernet Interface

This RJ45 socket has standard Ethernet pinout (for use of a normal “straight” cable to an Ethernet hub or switch). The interface normally operates in 100BaseT full-duplex mode, with auto negotiation, but it is possible to force other configurations.
The installation and operation of this function is detailed in 3.4, Initial setup of the Ethernet interface.

6.1.5. ISDN interfaces

The socket is ISDN S0 interface (BRI), the supported protocols are ETSI (Euro RNIS), NI-1 or NTT.

6.1.6. POTS/PSTN interface

Interface available as an option.

The telephone interface is a RJ11, only Tip and Ring signal are used.

Pin	Description
1	-
2	TIP
3	RING
4	-

6.1.7. Antenna socket (mobile networks)

Available on units equipped with the "wireless" option. See also chapter 2.1.4: Mobile network access.

This socket is used to connect one external (contact your local distributor for antenna accessory).

The interfaces are male HIROSE MS-151NB (female contact), 50 Ω impedance.

The internal mobile access module can operate in following bands:

Frequency band	850 MHz	900 MHz	1800 MHz	1900 MHz	2100 MHz
2G GSM/EDGE					
3G/3G+ UMTS/HSDPA/HSUPA/HSPA/HSPA+					

Specific versions exist for access to 4G/LTE networks (contact AETA).

6.1.8. Batteries supply

The unit can be powered by 6 type « C » or LR14 alkaline cells. Heavy-duty alkaline cells or rechargeable NiMh cells can be used.
Caution: OBSERVE THE RIGHT POLARITY WHEN INSERTING THE BATTERIES..

Only NiMh cells can be used due to the integrated charger in the unit. Please only consider alkaline cells as an emergency set. An automatic detection will avoid in such case the batteries to be charged

The autonomy depends on the selected algorithm and the network. With fully charged batteries, we have from 3 to 4 hours autonomy. A battery-like indicator displayed on the screen indicates the residual level of batteries.
Given that it is not usually possible to know how far a given set of batteries has been discharged before use, ensure that you recharge them after each broadcast.

IT IS STRONGLY RECOMMENDED THAT YOU REMOVE RECHARGEABLE BATTERY WHEN YOU DON'T USE THE SCOOPY+ FOR MANY WEEKS.

ⓘ *IT IS STRONGLY RECOMMENDED THAT YOU DON'T USE LOW QUALITY, SALT OR ALKALINE CELLS. SUCH BATTERIES MAY CAUSE LEAKS AND DAMAGE THE UNIT.*

6.1.9. DC power supply

Scoop+ will also work on any external 8 to 15-Volts DC source. A typical source will be a car cigarette adapter. Connect your DC power cord to the connector on rear panel of the unit (labelled DC 8-15V 2A), and connect the other end into your DC power source

The unit includes overvoltage and overcurrent protection (resettable fuse on the DC input).

The DC supply has priority on battery power

Characteristics	Min	Nominal	Max	Notes
Supply voltage	8 V	12 V	15 V	
Supply current	0.5 A		2 A	[1]

[1]: The power consumption may vary in a wide range depending on the input voltage, the installed options, and the operating mode.

6.2. Audio performance

The audio performance in this part applies to the system without coding/decoding, and excluding the coordination channel. The additional effect of the audio encoding and decoding on audio performance depends on the coding algorithm used and its parameters.

Except when differently stated, the following measurements are done at a +6 dBu input level and on the AD/DA path, with maximum input and output level set at +16 dBu.

6.2.1. Transmission gain

The drift in time of the gain from the input to the output of the codec is less than ± 0.3 dB.

6.2.2. Amplitude-frequency response

All measurements are done with a +6 dBu input signal, and a reference frequency of 1020 Hz. The measurements are done with a loopback before coding/decoding, so the possible effect of compression has no influence.

For a 48 kHz sampling rate:

Frequency range		Tolerance	
0 Hz	20 Hz	-∞	0 dB
20 Hz	40 Hz	-0.2 dB	0.1 dB
40 Hz	20 000 Hz	-0.1 dB	0.1 dB

For a 32 kHz sampling rate:

Frequency range		Tolerance	
0 Hz	20 Hz	-∞	0 dB
20 Hz	40 Hz	-0.2 dB	0.1 dB
40 Hz	15 000 Hz	-0.1 dB	0.1 dB

6.2.3. Group delay distortion

Taking the minimum group delay as reference, the group delay distortion on the AD/DA path is always less than 1 ms.

6.2.4. Idle channel noise

Background noise is measured with no audio modulation (idle channel), with maximum input and output level set at +16 dBu, through the whole encoder-decoder chain (wide band coding, with 48 or 32 kHz coding frequency).

Maximum noise level[1]: - 56 dBm (quasi-peak detection, CCIR weighting)
(or - 62 dBq0ps)

This result in a signal to noise ratio (SNR) of more than 72 dB.

When the maximum input and output level is set at another level, both the signal and noise levels are shifted but the SNR remains in the same range.

[1] Worst case for all types of algorithms; MPEG performs better than the others

6.2.5. Total distortion vs frequency and level

Total distortion relative to maximum level (or THD + N) is less than –78 dB over the whole audio bandwidth (20 – 20 000 Hz). This performance holds for audio signals from –80 dB to –1 dB relative to the maximum level (+16 dBu).

6.2.6. Crosstalk

Crosstalk is less than -80 dB over the whole bandwidth.

6.2.7. Gain and phase difference between channels

The gain difference between channels is less than ± 0.3 dB over the whole bandwidth, for any sampling frequency.

The phase difference between channels is less than ± 3 degrees over the whole bandwidth, for any sampling frequency.

6.3. Network protocols and ports

The SCOOPY+ implements or complies with the following protocols (non exhaustive list):

- Physical and link layers: Ethernet, 100BaseT, 10BaseT
- Network/Transport layers (IPv4): TCP/IP, UDP/IP, RTP/IP
- Application: HTTP, Telnet, DHCP, STUN, NTP, SYSLOG
- Audio transmission: SIP signalling, SDP, RTP, RTCP, RFC3550/3551, RFC3640
- Compliant with EBU recommendation Tech 3326 (interoperability of audio codecs for contribution)

The ports used by the device are the following:

Type	Port	Designation	Dir	Notes
TCP	80	HTTP	⇆	Embedded html server
	6000	Control	⇆	Remote control ("command line" mode); used by Scoop4Man and TeleScoop
UDP	123	NTP	⇆	For querying an NTP server
	514	SYSLOG	→	For sending messages to a SYSLOG server
	3478	STUN	⇆	For querying a STUN server
	5004	RTP	⇆	For audio streaming
	5005	RTCP	⇆	For audio streaming
	5060	SIP	⇆	SIP signalling
	6000	Multicast/description	⇆	Multicast stream description channel
	6001	Multicast/audio	⇆	Multicast stream transmission channel

*Ports shown in **bold** can be changed from the default values shown in the table.*

6.4. Dimensions and weight

Dimensions: (D x W x H) 234 x 155 x 80 mm
Weight: 1.5 kg, with batteries

6.5. Environmental characteristics

The equipment operates over a 0°C to 45°C ambient temperature range (32°F to 113°F), and a 5% to 90% humidity ratio range.

The SCOOPY+ complies with "CE" directives regarding safety and EMC.

- Safety: compliance with EN60950
- EMI: radiated emissions complying with EN55103-1
- Susceptibility: compliance with EN55103-2

6.6. Options

The basic version of SCOOPY+ includes Ethernet interface for IP transmission and/or Wireless module. Many available options can be added to complement the capabilities of the SCOOPY+.

6.6.1. Network options

For adding more types of network interfaces, the following options are available:

Ordering code	Option	Description
80 00 141 51	ISDN	Addition of one ISDN interface, for 64 or 128 kbit/s transmission (1 or 2 B channels)
80 00 142 51	POTS codec	Addition of the POTS/PSTN interface
80 00 149 51	Inmarsat BGAN	Direct control over an Inmarsat BGAN terminal, automatic use of streaming class.
80 00 143 51	Wireless option via USB (Scoopy+)	3G/3G+/LTE wireless with inserted module (not included)

6.6.2. Other options

Ordering code	Option	Description
80 00 145 51	AAC	Addition of AAC coding algorithms: AAC-LC, HE-AAC and HE-AAC v2
80 00 142 01	Record	Record and Edition of audio file
80 00 194 01	12V DC power	Additional 12V DC power supply input
80 00 053 01	Grounded Power supply	instead of standard one, recommended for some Sennheiser headset as HMD 24; HMD 25

6.7. Accessories and related products

The SCOOPY+is delivered with an AC/DC adapter and a CAT5 Ethernet cable.
The POTS option includes a RJ11 cable.

The "wireless" option is delivered with an internal multiband antenna
An adapter is available if needed for using a µSIM card instead of a standard SIM card.

SCOOPY+ can also be controlled by several third-party software tools such as Codec Live, MDC.Net...

For an additional Ethernet interface, usable for remote control from a separate sub-network, it is possible to plug on the USB socket a USB/Ethernet adapter, such as the LogiLink UA0144.

Other compatible devices may be available later, in which case you can find them on our web pages; consult our site www.aeta-audio.com .

80 00 146 01	Dedicated bag for Scoopy+	...
ZAAS00040	External wireless antenna	Optional complement for ".../Wireless" versions
80 00 147 01	Rechargeable Ni-MH battery set	Set of Ni-MH batteries, 2500 mA.h
80 00 030 01	Car battery adapter	With cable for Scoopy/Scoopy+/Scoopfone HD DC input, 3 m
80 00 017 11	External Power Adapter	Universal mains input
ZAAS00055	GPIO extension box	Adds to a Scoopy+ 4 opto-isolated inputs and 4 relay outputs
ZAAS00056	µSIM adapter	Allows to use a µSIM format card (for wireless versions/options)
80 00 151 01	Scoopy+/Mixy coupling cable	For using Mixy as a 3 additional inputs extension on a Scoopy+
15 00 073 31	Grounded Power supply	Grounded power supply recommended for some Sennheiser headset as HMD 24; HMD 25

Accessories (Contact your distributor for more accessories)

• Carrying bag

• Car DC adapter cable

• Spare AC/DC adapter

• External multiband antenna

• Audio extension: MIXY

7. Annexes

7.1. Additional information on the algorithms and protocols used

7.1.1. Auxiliary data in the MPEG frames

The auxiliary data are used for the following purposes:

- Reed-Solomon error detection and correction (J52 standard)
- Data channel
- Other auxiliary information: relay transmission, and/or auxiliary audio channel. The insertion of this auxiliary information is an extension (AETA proprietary format) to MPEG. However, the frame structure remains compliant.

7.1.2. Reed-Solomon encoding

In order to cope with possible transmission errors in the network, Reed-Solomon error correction coding can be added, compliant with J52 recommendation. Four correction modes are available in the SCOOPY+:

- Mode 0: no error correction, Reed-Solomon coding disabled
- Mode 1: protection of only the control information and scale factors in the MPEG frame, low redundancy
- Mode 2: protection of the whole frame, moderate (2.5 %) redundancy
- Mode 3: protection of the whole frame, high (10 %) redundancy

Higher redundancy increases the protection against errors, but slightly degrades the audio quality, as redundancy takes up part of the bit rate that could be allocated to audio coding.

Most often, for a normal quality transmission link, mode 1 is sufficient and it consumes little bit rate from the compressed data, so it hardly impacts the audio quality. Although J52 does not apply to leased line connections, this error protection technique is also implemented in the SCOOPY+ in leased line mode.

7.1.3. H221 framing

H221 defines a framing structure that allows byte synchronisation recovery in leased lines, and the transmission of control data along with the main data.

7.2. Overview of the SIP protocol

7.2.1. What is SIP?

SIP is for Session Initiation Protocol, a protocol specified by the IETF for establishing media transmission sessions. SIP is considered the communication protocol of the future by most vendors, and as such it has deep influence on the VoIP applications.

As a signalling protocol, SIP brings methods and techniques to solve the issues related to the establishing of an audio link. Almost as important, it is a recognised standard, implemented on many network devices and systems. Using SIP helps you build modular and really evolutive systems, not being tied to a single vendor.

The SIP protocol is an essential requirement in the EBU Tech 3326 recommendation (a.k.a. "N/ACIP" recommendation, from the name of the EBU workgroup who elaborated it).

7.2.2. Setting a link with SIP

Let us look at an example (diagram below): a reporter on the move with a Scoopy+[1] wants to make a call to a SIP compliant codec located in the main station. The reporter may be at home, or at another location, not necessarily known in advance.

Once the Scoopy+ is on and connected to the network, it will register itself ❶ to a SIP "registrar". This registrar can be located on the LAN of the radio house, but it may as well be elsewhere in the network. Then the registrar "knows" where the Scoopy+ is, what its IP address is. On the radio house side, a similar process takes place ❷.

To call the codec of the radio house (e.g. a SCOOPY+), the reporter just needs to know its SIP address, which can be like studio12cod@radiomcr.com (indeed very similar to an e-mail address). To call the unit, the reporter has to select the preferred audio coding mode on the Scoopy+ (e.g. mono G722), then call the remote unit, simply using this SIP address (SIP URI).

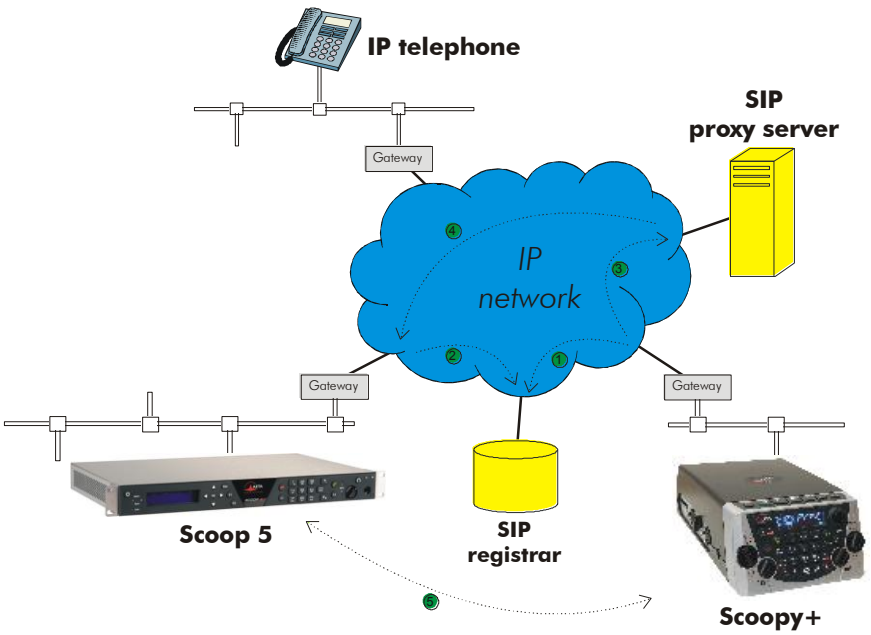


Figure 9 – Setting up an SIP session with server(s)

[1] Scoopy+ is a portable audio codec from AETA; the description here applies to both Scoopy+ and the Scoop 5, as they are both SIP compliant and mutually compatible

What happens then on the network: the Scoopy+ sends the request ❸ (INVITE in SIP protocol) to a proxy server (often the same device is also the registrar). To make things simple, this proxy then relays and routes the request ❹ to its destination. Resolving the SIP URI to a physical network and address uses mechanisms similar to those used for resolving URLs. Several proxys may be involved in cascade to finally reach the desired destination, but this does not have to be known or dealt with by the end devices. The following is like the initiation of a phone call: the IP codec "rings"; if it accepts the call, this is notified to the Scoopy+.

At this stage, the proxy (s) provide the Scoopy+ and the IP codec with all the address data they need for the link, then the actual audio streams can be exchanged ❺ between both units. As a very important feature, the end devices now can exchange data directly; the proxies do not have to be on the path, they are only involved in the setting (and later the ending!) of the session. The codecs will automatically exchange their coding capabilities, and agree on a coding mode, with no further user intervention.

Alternatively, the call can be done from the station to the reporter, in a way very similar to the above. In contrast with ISDN links, the operators at the station do not even need to know where the reporter is located! This is because the registrar deals with this issue.

Note that it is also possible to set a link with a SIP-compliant VoIP phone instead of another codec. This is one of the numerous advantages of using a standard.

7.2.3. Setting a link without a SIP server

Contrary to a commonplace but wrong idea, the SIP protocol does not impose using a SIP server. The SIP *protocol* can be used without a SIP *server*, i.e. it is possible to set "peer to peer" links without involving one or more SIP *registrars* or *proxies*. In such a case the process is more direct, and the codec contacts the remote destination with no intermediate entity. However, there are some drawbacks with such type of session:

- Without a *registrar*, the identification of the destination is its IP address; this is not as "stable" as a SIP URI, as it may change depending on the location, or time (with dynamic addressing).
- The two codecs must have open access to the Internet, or otherwise the link must be made possible by unlocking if necessary the appropriate routes and ports (conversely, when using a proxy, such allowances can be restricted to the server connection, for a better control on the access security).
- Most often the gateways perform address translation (NAT), which is an *a priori* issue for UDP protocols such as implemented for real time audio links. Proxies can help dealing with this issue, but without such servers it may be rather difficult to work around such obstacle. *The following chapter provides some recommendations about this.*

7.3. Some methods to deal with NAT routers and firewalls

Problems arise when the desired connection has to go through a NAT router, and/or a firewall, that blocks a direct IP communication.

This is a very common issue, especially if one needs to set up a transfer via the Internet. It is impossible here to describe in details the possible ways to deal with this problem, but the following just shortly discusses some typical solutions. Most probably, a network administrator should be consulted for support, and for granting adequate network authorisations and/or privileges.

The most classical issues are related to:

- The presence of a NAT router on the network path between the codecs.
- The presence of a firewall on this path.

It is always important to have available the information regarding the network organization and to be allowed to access the devices which need to be configured. Hence we highly recommend to involve the persons empowered for such tasks.

7.3.1. Links via a private network

No special problem should be met within a LAN. The operation is also possible with codecs from other manufacturers, provided that they comply with the Tech3326 EBU recommendation (also known as "N/ACIP" recommendation). However you should check for specific settings or preparation possibly needed on such devices.

A wide area network covers a wider geographic range, and the network topology most probably includes routers on the path between the codecs to be linked. However, usually there is not much difference with a local area network.

Note: using a VPN leads to just the same case; the operation is identical as far as the codecs are concerned.

7.3.2. Links through a public network (Internet)

If each of the units has got a "direct" access to Internet with a public address, we are in the same situation as the previous one, functionally speaking (private WAN). The addressing scheme is normally static, as DHCP can rarely be used on a public access. In fact, this situation is very seldom met in the field!

First, the Internet access is usually protected by a firewall which will, as a principle, block a priori the desired connection. In such case exceptions (to the firewall security rules) must be created, that will allow this connection; this has to be done by the person in charge of the network management.

Most often, on one access if not both, the codec accesses the Internet via a NAT router. This router shares Internet access, with one or a few public addresses, among the equipment on the LAN. On this LAN the devices get local private addresses, and the router carries out an IP address translation. Note that:

- As an example, a consumer ADSL modem-router is almost always a NAT router, sharing a single public IP address between the devices connected to the router.
- It is just the same on a 3G/3G+ mobile IP access; the terminals (phones or computers) access the Internet via NAT routing.
- NAT routing is often included in the firewall features; in fact NAT routing somewhat participates to the protection against direct attacks from the outside.

NAT routing is an obstacle to transmission with UDP, mainly for two reasons:

- It does not allow unsolicited data to come in from the outside network. In other words, data input is accepted on a port as an answer to a request from the local network, but an external agent cannot directly initiate the transmission of a packet.
- The terminal units on the LAN only know their private local address. On the other hand, agents implementing the SIP protocol have to communicate to each other the addresses and ports to be used for the media exchanges. Because of the NAT routing, agents do not get the real public addresses, which leads to failure of the session setup attempts.

We are now looking at various methods used to overcome these obstacles.

NAT and use of a STUN server

The STUN protocol is a method which is often successful[1] in helping the agents to discover their public address even when they are "hidden" behind a NAT router. Here is the operation principle:

- A STUN server is used, which is accessible over the Internet;
- The address of this server is programmed into the agent (i.e. the audio codec in our topic);
- The agent queries the server and discovers its public IP address and port number, as seen from outside of the NAT router and LAN;
- This addressing information is then used by the agent for negotiating and setting up a media session.

The STUN server address is programmable in the menu or the html pages of a SCOOPY+ or Scoopy+. Besides, there is also in the menu (keypad and display on the front of the unit) an enable/disable (on/off) selection, without having to clear the server address.

There are many public STUN servers available on the Internet; here are a few examples, valid at the time of writing:

Domain name	Numeric address
stun.aeta-audio.com	**85.214.134.163**
stun.ekiga.net	77.72.174.163
stun.sipgate.net	217.10.68.152

Examples of STUN servers

It is recommended to check that the server is operative. Moreover, numeric addresses may change, even if the domain name stays the same. A list of servers can also be found on the support page of our web site http://www.aeta-audio.com.

[1] Although not with so called "symmetric" NAT routers

Standard NAT router

Situation: codec A behind a NAT router with no specific programming *(a codec accessing Internet via a mobile network is almost always in such situation)*.

Besides, we assume that the other codec (B) is accessible with a public address.

Once codec A is configured for using a STUN server:

- codec A can initiate a connection to (call) codec B
- codec B cannot call codec A

Advantages	Drawbacks
Configuration is relatively simple	B cannot call A
No change is needed on the router	
Several codecs can be set behind the NAT router	
Method suitable for mobile network access[1]	

NAT router with DMZ

Situation: codec A behind a NAT router and placed in « DMZ ».

We also assume that the other codec (B) is accessible with a public address.

Once codec A is configured for using a STUN server:

- codec A can initiate a connection to (call) codec B
- codec B can call codec A, using the public address of the NAT router

Advantages	Drawbacks
Each codec can set up a session	Need to configure the router
A is nearly equivalent to a codec with a direct public access	Only one codec can be set up in this way on a LAN
	A is exposed to external attacks
	The DMZ may be already reserved for other network equipment
	Method not possible for a mobile network access

NAT router with port forwarding

Situation: codec A behind a NAT router and configuration of the router to forward to A the necessary ports.

We also assume that the other codec (B) is accessible with a public address.

Port forwarding to be set on the router[2]:

- UDP 5060 (= SIP port)
- UDP 5004 (RTP port) and 5005 (RTCP port)

[1] Except with symmetric NAT, which is often met for mobile network access

[2] If needed you can change these numbers on the SCOOP 5

Once codec A is configured for using a STUN server:

- codec A can initiate a connection to (call) codec B
- codec B can call codec A, using the public address of the NAT router

Advantages	Drawbacks
Each codec can set up a session	Need to configure the router
A is nearly equivalent to a codec with a direct public access	Only one codec can be set up in this way on a LAN
	Method not possible for a mobile network access

Use of a SIP server

In addition to the numerous features it brings along, using a SIP proxy server is a very powerful method to solve the issues related to NAT routers, because most SIP proxies are capable to detect the presence of NAT routers and/or deal appropriately with their traversal.

If a SIP server is available, and once the codecs are registered on this server:

- Any registered codec can call another registered[1] codec, regardless whether there is or not a NAT router on the path.
- The identifier (SIP URI) is stable and does not depend on the location of the called agent ("mobility" feature).

It is possible either to use a public server on the Internet, or to install a private server accessible via the Internet.

Advantages	Drawbacks
Each codec can initiate a session Each codec can receive calls	Installation may not be easy (private server)
Identification is simple and location/time-wise stable	Reliability of server questionable (public server)
Security: a private proxy can be linked with a firewall	
Also works with symmetric NAT routers	
Interoperation with telephony over IP	
Method suitable for mobile network access	

For a fast implementation, you can use SIP accounts on AETA's SIP server sip.aeta-audio.com. This server is dedicated to a professional broadcast usage, and housed in a safe site, available 24/7. Contact AETA for subscribing SIP accounts.

[1] Depending on the access control policy, a server may accept « outgoing » calls to third party domains, or accept « incoming » calls from non registered agents.

7.3.3. Summary and reminder of essential rules

The table below sums up the situations where a link can be set up (not using a SIP proxy server) and reminds the needed specific settings:

	Codec A access	Possible calls	Codec B access	Notes
1	LAN	⇒ ⇐	LAN (same)	
2	Private WAN	⇒ ⇐	Private WAN	
3	Internet direct	⇒ ⇐	Internet direct	
4	NAT	⇒	Internet direct	STUN needed for A
5	NAT + DMZ	⇒ ⇐	Internet direct	STUN needed for A
6	NAT + port forwarding	⇒ ⇐	Internet direct	STUN needed for A UDP ports 5004, 5005, 5060
7	NAT	⇒	NAT + DMZ	STUN needed for A and B
8	NAT + DMZ	⇒ ⇐	NAT + DMZ	STUN needed for A and B
9	NAT + port forwarding	⇒ ⇐	NAT + DMZ	STUN needed for A and B UDP ports 5004, 5005, 5060
10	NAT	⇒	NAT + forwarding	STUN needed for A and B
11	NAT + DMZ	⇒ ⇐	NAT + forwarding	STUN needed for A and B
12	NAT + port forwarding	⇒ ⇐	NAT + forwarding	STUN needed for A and B UDP ports 5004, 5005, 5060

ⓘ *Basic rule: Codec behind a NAT router => use a STUN server.*

This allows the codec to set up *outgoing* calls. This is not sufficient to be accessible to connection requests *from* the outside.

ⓘ *Mobile network access without SIP server or VPN => use a STUN server*

ⓘ *NAT + DMZ or NAT + forwarding => incoming calls are possible.*

Incoming calls are not possible behind a NAT router without either such change or a SIP proxy.

ⓘ *SIP server => maximum versatility, at the expense of some initial effort (for installation)*

ⓘ *Reminder: the SIP protocol (always used by AETA codecs) does not impose the use of a SIP server. Codecs can set up point-to-point links using this protocol in the above described conditions. When no SIP registrar is involved, the identifiers are simply the IP addresses of the codecs.*

7.4. Notice regarding open source code

The software of this product includes programs and libraries that are covered by the GNU General Public License (or "GPL"), available for example at following address: http://www.gnu.org/licenses/gpl.txt . Under this license, the source code for concerned elements is available on our Internet site ("Download" page); otherwise it can be obtained on request by e-mailing AETA AUDIO SYSTEMS (open_source@aeta-audio.com).

The software also includes the sipX stack covered by the LGPL license.

8. Index

4

4SB, ADPCM 15, 25

5

5AS 1, 9, 59

A

AAC 16, 104
ADPCM 15, 25
Antenna 11, 34
Audio Extension 19, 34
Auto redial 59, 92
Auxiliary functions 91

B

BRI 61

C

Clock 9
Consumption (power) 100
Coordination channel 19, 26, 59, 91
Crosstalk 102

D

Data channel 25, 59, 107
DHCP 36
Distortion 102
Double codec 13, 63, 83

E

EMC, EMI 103
Environment 103

F

Factory settings 35, 77
Firewall 110

G

G711 14
G722 ..... 15, 25, 108
GPL, LGPL 115
Group address 67

H

H.R. 103
H221 ..... 9, 15, 107
H242 ..... 9, 15
HTML server 80

I

Impedance 98
International sound 18

J

J52 ..... 9, 15, 107

L

Local address 61
Loop
  Loop control 92
  Test loop 28

M

MPEG AAC 16, 104
MPEG Audio Layer 2 16
Multicast 67, 86

N

NAT 46, 110
NI-1 (protocol) 9
NTT (protocol) 9

P

Packet replication 8, 47
Password 35, 40, 75, 77
PIN code 51, 69, 71, 88
Port (forwarding) 112
Port(s) (TCP/UDP) 47, 103
Profile 27, 56, 74, 84
Protocol 103
Proxy 109

R

Record
  Routing 19
  Use 78
Redial (auto) 38, 59
Relays 25, 107
Remote control 8, 29, 36
Replication (packet) 8, 47
Restricted access 40, 60, 75

S

SD card 19, 34
SDP 8
Send FTP 79
Server 113
SIM card 11, 34, 51, 69, 71, 105
SIP
  protocol 7, 8, 108
  registration 108
  server 109
  URI 65, 108
SNR (Signal to Noise Ratio) 101
Sound Card(USB) 19, 34
SPID (Service Profile Identifier) 61
SRT 15

Start up 40
STUN 46, 103, 111
Sub-address 42, 61, 62
Supervision 27, 29
Synchronisation 9, 15

T

TDAC 15
Tech 3326 (N/ACIP) 103, 108
Temperature 103
Test loop 39
THD, THD+N 102

U

Unicast 7
USB 12, 34, 44, 51, 71, 88

V

VoIP 108
VPN 110

NOTES

AETA

AUDIO SYSTEMS ®

superior sound since 1978

AETA AUDIO SYSTEMS S.A.S.
IMMEUBLE KEPLER 4 – PARC TECHNOLOGIQUE
18-22, AVENUE EDOUARD HERRIOT
92350 LE PLESSIS ROBINSON – FRANCE
TÉL. +33 1 41 36 12 00 – FAX +33 1 41 36 12 69
http://www.aeta-audio.com

Specifications subject to change – All rights reserved by AETA AUDIO SYSTEMS

55 000 059 - C December 2013